The American University Laboratories For Electrical Engineering Part 1

Conference Paper · May 2018

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Engineering students will find this book very interesting. The book is documentation of Labs for engineering. It is facts that are important to design systems. All computer work and results and programs are given. simulation software are used and results are explained. The Labs are electronic systems and the results. comparing results with theory is done. As a result the student will feel theory and practice.

I am in Boston at Northeastern University for 8 years. This work is documentation of some of the things I did. A student doing electrical and computer engineering will find this book very interesting.

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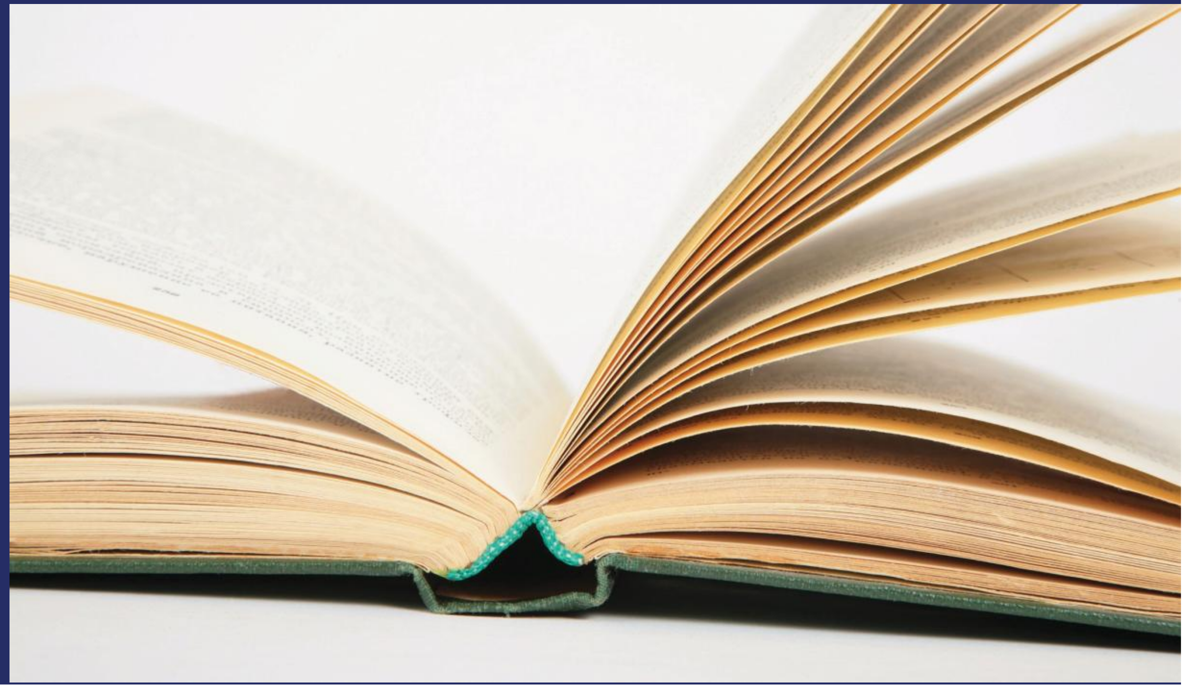
Labs for electrical engineering

Sobih



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The American University Laboratories For Electrical Engineering

Part 1

12/20/2012

Dr. Ziad Sobih

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To my two sons

Abdulrahman and

Kareem

Content

Electrical engineering

Electrical engineering can be divided into five parts. First is power engineering. Second is computer engineering. Third is electromagnetic. Forth is signals and systems. Fifth is electronic and circuits.

First power engineering, where we study generation and transmission of electrical power. Machines such as motors and generators are studied. Power plants where fuel is converted to mechanical energy and then to electrical energy are studied. Transmission lines are studied. In this part the energy conversion law is needed.

Second is computer engineering. In this part we start by logic and minimizing the number of gates. Software at low level language will be developed. Sequential machines will be studied. Memory systems will be studied. Microprocessor hardware will be developed. Programming microprocessor systems will be done.

Third is electromagnetic. Maxwell equations will be introduced. Uniform plane waves will be studied. Boundary conditions will be studied. Waveguide and transmission line systems will be studied.

Fourth are signals and systems. This has control systems and communication systems and digital signal processing. Control system use feedback with a gain k to control a planet. Changing the gain k changes the location of the poles to the desired specifications. Communication system such as AM and FM are studied. Digital signal processing filters are to be designed.

Fifth are circuits and electronics. We develop fast ways to solve networks and the use of transistor as amplifier and the use of transistor as a switch.

The low pass filter

We have a square wave as an input to a system of a low pass filter. The low pass filter is a resistance and capacitance in series. At the output across the Cap the signal will start to increase as a response of a first order system of a unit step input. This low pass filter is a first order differential equation. The Impulse response is an exponentially damped wave.

If we take the differential equation of this system to the frequency domain and solve it, we will find that it has one pole on the negative of the real axis. At this point we want to calculate the frequency response. As we move along the jw axis the distance to the pole will increase and because we are dividing by that distance the frequency response will get lower.

An important value of interest to us is the 3 db point and this will happen when we divide by 2^1/2. This is the half power point because the power is related to the square.

At this point, we have a right angle triangle one side is the pole at the negative real axis and the other side is the same distance of the pole at the jw axis. The distance that we have to divide by to get the frequency response is 2^1/2 times the pole value.

This means that when we are at the pole distance from the zero on the jw axis we are at the half power point.

For any linear system to calculate the frequency response we move along the jw axis and we calculate the distance from the frequency point at the jw axis to all of the zeros and the distance to all poles. Then we multiply all the zero distances and divide by the pole distances. This will give us the steady state gain at that frequency.

This will give us an idea about the design for a system where we want a max at some frequencies and a min at some places. We can play with the zero-pole map using this rule to meet our design specifications.

If the input to the low pass filter was a sin wave the output is also a sin wave with a gain and a phase shift. This is because it is a linear system. At low frequency the gain is one and the phase shift is zero and we can see this for the RC circuit if we play the input and the output on the scope. As we increase the frequency until we are at f=1/RC we get the half power point where the gain is .707 and the phase shift is −45 degrees. As we increase the frequency the gain will go down more and more and the phase shift will go down to −90 degrees. All of this can be monitored on the scope.

Let us give the rule in a linear system for the phase shift. We have to sum all angles from the frequency on the jw axis to the zeros minus all angles from the same point to the poles. The input is a sin wave and the steady state output is a sin wave of the same frequency with a gain and a phase shift.

Digital System First Order

Sampling Theory is very important in Digital signal processing. If we have a signal with a highest frequency F, sampling theory says that we have to sample at 2*F or more in order not to loss any information. If we do that we can reconstruct the original signal from the samples with no loss.

The difference equation is one way to describe a digital system [17]. For example

$$y(n) = x(n) - a*y(n-1)$$

This is a digital low pass filter with time constant equal to RC. By that I mean if we have an input x(t) and we sample it above twice the max frequency in x(t) so we get x(n). y(n) is the output of the system using the difference equation of the low pass filter system. Then we reconstruct the output to analog signal using the reconstruction formula. We will get a low pass of the input with a cutoff frequency equal to 1/RC. I use this form for the equation to make it easy to synthesize and simulate using MATLAB.

For a digital system we need analog to digital converter which will sample the input signal. Then we process the samples according to the difference equation. The output is given to a digital to analog converter.

We can have an idea of the design if we look at the z transform. As we move around the unit circle the distance to the pole of the low pass filter will increase for the frequencies from zero to Fs/2. Because we divide by this distance the frequency response will decrease. The pole is at the real positive axis inside the unit circle. When we increase the frequency from zero to Fs/2 we cover the upper half of the unit circle.

We can experiment this in a lab by having a microprocessor board and a timing chip. The timing chip will interrupt every Ts second giving order to sample the input. Then the microprocessor will take the new sample and do the calculation according to the difference equation of the low pass filter. The output will be given to a digital to analog converter.

If we do this work and the input and output are played on an oscilloscope and as long as we do not violate sampling theory we will find that the output is a low pass version of the input.

As we change the place of the pole we change the place of the cutoff frequency

Second Order System

Some systems in nature satisfy second order differential equation. This system might be mechanical or electrical. An example for mechanical system is the pendulum. In this system if there was no friction and the angle was small the second derivative of the function is approximately equal to the function. And the solution to such case is a simple harmonic motion.

The electrical system dual to the pendulum is the LC circuit. If there was no resistance we have a similar equation and the same solution.

When we have a resistance we will have an exponential damping to the sin wave which is the response of the impulse. All of this can be shown using math. The transfer function of the second order system has two poles which are the roots of the quadratic equation. The two roots have negative real part if the system is stable and if there is an imaginary part they will be in complex conjugate. The real part is a measure of the exponential decay. The imaginary part is a measure of the frequency of oscillation.

We will have the fastest step response when the real part and the imaginary part are approximately the same. Using these facts we can design to meet the needed specification.

Many parameters are important to control system design. For example the time the step response rest to constant or the transit time. And the percentage overshoot and how fast the response is?

All these parameters can be related to the location of the poles and the poles can be adjusted to meet desired specifications. For example increasing the absolute value of the negative real part of the pole will make the step response faster and the transit time smaller.

A pendulum motion can be simulated using the duality between mechanical and electrical system. Electrical systems are easy to simulate using computer software. For example the friction force is dual to the resistance. And the second order differential equations that govern RLC circuit are similar to the equations that govern the pendulum.

We can use resistance, capacitance and inductance to have the same coefficient a, b and c for the second order equation for both the mechanical system and electrical system. And as long as the model we use to best describe the system is the same in both electrical and mechanical systems, we can use duality to simulate or solve the math.

Digital Second Order System

The difference equation for a second order digital system [18]

$$a*y(n)= x(n)-b*y(n-1)-c*y(n-2)$$

As we can see there is unit delay and two units delay for the output that is used in feedback. If we sample at a rate Ts, the max frequency that we can work with is 1/(2*Ts). We can input to the system a unit pulse and look at the output or a unit step and look at the output.

The output can be calculated by using a computer and program it to use this equation.

The equation is in this form to make it easy to synthesize the system.

If we take the z transform of this equation and solve for the system transfer function, we will get two poles inside the unit circle if the system was stable. They might be in complex conjugate. The places of the poles determine the frequency response and the time domain response.

To design for certain specification in the time domain or in the frequency domain we place these poles in the right place. For example the closer the pole to the unit circle the longer the transit is.

The same rule for the analog system can be used for the digital system. I mean the transfer function magnitude can be calculated for a frequency on the unit circle by multiplying the distance to all zeros and dividing by the distance to all poles. The distance we use is the distance from the frequency on the unit circle to the pole or zero.

For a linear system the transfer function has a magnitude and a phase for each frequency. The steady state response for a sin wave input will have a gain equal to this magnitude to the input and the input will be shifted by this phase.

The percentage overshoot for the step response and the time to rest to a constant and the speed of the response and its natural frequency can be related to the location of the poles and zeros. All of the quantities are in the time domain.

In the frequency domain the max of the transfer function and the min can be related to the location of the poles. For example if we want a max at a certain frequency we place a pole close to that frequency on the unit circle. If we want a min we place a zero. So we can place the zeros and poles in our design to meet the desired specifications.

Electromagnetic

Vectors and Phasors

A sin wave can be described completely by an amplitude and phase and frequency. For a linear system in steady state the frequency of all inputs and all outputs are the same. Using this fact in the system we can use only the amplitude and phase to describe the inputs and outputs.

For example if we have two sin waves of the same frequency. The first can be completely described by a vector with amplitude and phase and the second can be completely described by its amplitude and its phase and the sum is the vector sum of these two vectors. The result is a new vector with an amplitude and phase of the total.

A cos wave can be represented as a vector from the origin with magnitude of the cos wave amplitude and a phase angle from the positive real axis equal to the cos wave phase. As this phasor rotate with the speed of the frequency of the cos wave, the projection on the x axis is the cos wave. The motion on the x axis is the cos wave and we can see that if we give it time dimension.

In the linear system we add cos waves and we add first derivative and second derivative. The derivative of sin is cos and the derivative of cos is −sin. As we can see this is a phase shift and multiply by a scalar due to the chain rule. The point is the frequency does not change. This fact is used to solve a phasor equation for a linear system.

Please note that this work for addition of sin waves and addition of their derivatives and does not work for multiplication of sin waves. If we multiply sin by sin the frequency will change and as a result we cannot use phasors.

The picture that I want to make clear that in steady state of a linear system every phasor is rotating with the same speed and we want to add these phasors. These phasors are the scaled function and the scaled derivative and the scaled second derivative for the second order differential equation. The total is a phasor that rotate with the same speed. To get this phasor we freeze time and get the total sum of a function and the first derivative and the second derivative scaled as in the second order differential equation. The vector that we will get is the phasor of the output as in the equation.

Phasors are a good way to describe and solve linear differential equations which are used as a model to many physical systems in nature.

Maxwell equations

Maxwell four equations summarized electromagnetic the way Newton three laws summarized mechanics. Reading Maxwell equations and understanding them is good for the soul. Einstein a great admirer of Maxwell once wrote of him “Imagine his feeling when the differential equation he had formulated proofed to him that light is a form of electromagnetic wave that travel with a velocity of c“.

The first Maxwell equation is Faraday law. The second is Ampere law modified by Maxwell. The third is Gauss law electric. The fourth is Gauss law magnetic.

The best way to describe the physics is the integral form of the equation. However it is may be ten times harder to solve integral equation than differential equation. That is why we change the integral form to differential one.

To do this we introduce the curl of a vector and the divergence. By definition the curl is circulation over area and the divergence is flux over volume.

We will start by integral form of each law to give some insight of the physical meaning. Then we are going to convert the integral form to a differential one.

Faraday Law

Faraday was working in his lab in London when he realized that moving a magnet near a coil will generate voltage. Faraday law say's a voltage will be generated if the magnetic flux is changed through a wire loop. This was the concept to make generators where mechanical energy is converted to electrical.

The mathematical description of the law is the integration of the electric field around a closed loop which is voltage is equal to the derivative of the magnetic flux with respect to time.

But how did Faraday find his law? At that time a compass with some terns of isolated wire around it was used to measure current. The ends of the wire was connected an inductor. When faraday puts a magnet in the inductor, the needle of the compass turns meaning that there is current in the system with no battery connected.

Also the opposite is true. If an inductor is connected to a battery it will be a magnet. Faraday was working in his lab when he made all of these observations by chance. His law was a fundamental law in electromagnetic theory. It is one of Maxwell four equations that summarize electromagnetic theory.

To solve Maxwell equations we have to put them in differential form. Circulation over area is the curl. Let us take Faraday law for a small area and divide by this area. As a result we will have curl of E equal to the derivative of the magnetic field density

Curl (E) = dB/dt

If we have a rectangle delta x and delta y in the x−y plane and we apply to it Faraday law and divide by the area, we will have

$$((Ex+)dx-(Ey+)dy-(Ex-)dx+(Ey-)dy)/(dx)(dy)=-jwHz/(dx)(dy)$$

$$dEx/dy - dEy/dx = -jwBz$$

$$\text{curl } E(x,y) = -jwBz$$

This is for E in two dimensions. In space E might have three dimensions. So we have to do the same thing for the x−z plane and the y−z plane. For E in the x−z plane B in the y direction. For E in the y−z plane B in the x direction. We use the right hand rule to get the direction. If we do the math we will have

$$\text{Curl } E(x,y,z) = -jwB(x,y,z)$$

Ampere Law

Ampere realized that when a current pass through a wire it will have a magnetic field around it. Electricity and magnetism were two different subjects and his experiment showed that they are related. The magnetic field was felt by a compass near the wire.

He invented a way to measure current using his law. Some turns of wire around a compass were used to measure current.

Maxwell using symmetry added the displacement current to ampere law. With this addition he was able to solve the equations and predict the electromagnetic spectrum from pure math.

To understand the displacement current think of a wire in the y axis and a parallel plate capacitor connected to it at the origin. If we have a current I in the wire the rate of change in the electric field in the capacitor is proportional to the current. So we have a current and a displacement current which is proportional to the rate of change of the electric field.

At this point we have the second Maxwell equation. That is the closed integration of the magnetic field around a loop is equal

to the total current out of the loop plus the displacement current which is the rate of change in the electric field flux.

To make it easier to solve Maxwell equations we have to put the equation in the differential form. Circulation of a small area over this area is curl. If we divide both sides by small area and make the circulation over this area we will have curl of H and this is equal to current density plus the rate of change of electric field density with respect to time.

Curl H = j + jwD

To convert the integral form to differential form we take a small rectangle delta x and delta y and do the integral and divide by this area

(Hx+)dx − (Hy+)dy −(Hx−)dx +(Hy−)dy/dydx =Iz/dydx−jwEz/dydx

dHx/dy −dHy/dx =Jz−jwDz

curl H(x,y) = Jz − jwDz

This is for H in the x−y plane. For H in space we have to do also the x−z plane and the y−z plane. The result is

curl H (x,y,z) =J (x,y,z)+jwD(x,y,z)

Gauss Law Electric

The third Maxwell equation is Gauss law electric. If we have a point positive charge the electric field is out of this point in all directions. Spheres around this point where this point is the center are equal potential surfaces. The field is perpendicular to the equal potential surfaces. So we can say that the equal potential surface is a gauss surface where we can multiply the field by the surface to get the flux.

Gauss law electric say that the electric flux in a gauss surface is equal to the charge enclosed. Coulomb's law is a special case of gauss law. We chose the gauss surface so that the integral is easy to do. For example for a point charge we chose the gauss surface as a sphere around it.

E * (Area of the sphere around q) = q/ epsilon

This is Coulomb law derived from gauss law. This law is very important to calculate the energy of a system of charges. If we have E we can get the force and if we have the force we can get the work.

The integral form of Gauss law electric is that the surface integral of the electric flux is equal to the charge enclosed. We have to take it to the differential form to be able to solve Maxwell equation much easier.

In calculus flux over volume is the divergence. If we take the flux over a small volume and we dived by it we will get the divergence.

If we have a small cube delta x and delta y and delta z, the flux over volume if we have only Ex is

(Dx+) dy dz −(Dx−) dy dz / dx dy dz = Q / dx dy dz

dDx/dx =rou

Div D = rou

This is for Ex only. We have to do this for Ey and Ez also. The result is

dDx/dx + dDy/dy + dDz/dz = rou

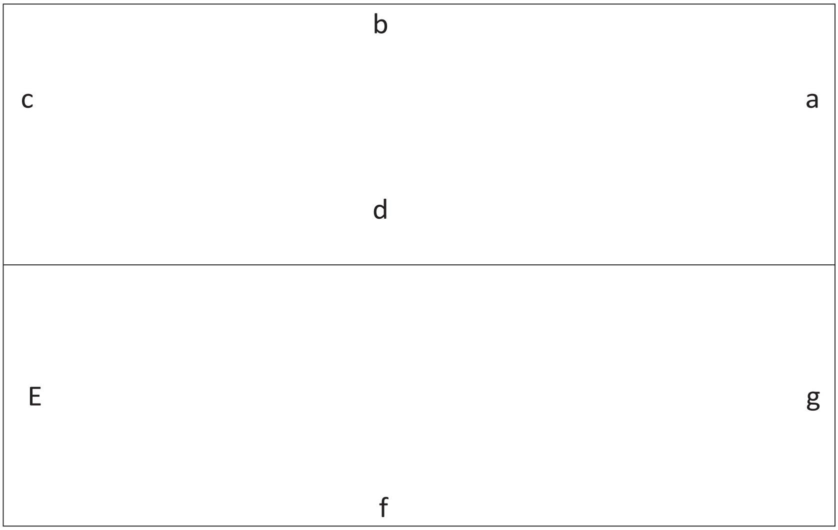
Div D = rou

Digital Systems

Seven Segments Display

The seven segments display is a way to show numbers from zero to nine. The numbers from 0 to 9 are in binary system. We use logic to go from binary system to show on the seven segments display.

a	b	c	d	e	f	g	Binary Number
1	1	1	0	1	1	1	0000
1	0	0	0	0	0	1	0001
1	1	0	1	1	1	0	0010
1	1	0	1	0	1	1	0011
1	0	1	1	0	0	1	0100
0	1	1	1	0	1	1	0101
0	1	1	1	1	1	1	0110
1	1	0	0	0	0	1	0111
1	1	1	1	1	1	1	1000
1	1	1	1	0	1	1	1001



This is the seven segments display and the table to link between the binary number and the display.

Dz/yx	00	01	11	10
00	1	1	1	1
01	1	0	1	0
11	x	x	x	X
10	1	1	x	X

A=d+z(not)+x(not)y(not)+xy

Dz/yx	00	01	11	10
00	1	0	1	1
01	0	1	1	1
11	x	x	x	X
10	1	1	x	X

B=y+d+zx+z(not)x(not)

Dz/yx	00	10	11	01
00	1	0	0	0
01	1	1	0	1
11	X	x	x	X
10	1	1	x	X

C=d+x(not)z+zy(not)+x(not)y(not)

Dz/yx	00	01	11	10
00	0	0	1	1
01	1	1	0	1
11	x	x	x	X
10	1	1	x	X

D=d+yx(not)+yz(not)+y(not)z

Dz/yx	00	01	11	10
00	1	0	0	1
01	0	0	0	1
11	x	x	x	X
10	1	0	x	X

E=z(not)x(not)+yx(not)

Dz/yx	00	10	11	01
00	1	0	1	1
10	0	1	0	1
11	X	x	x	X
01	1	1	x	X

F=z+d(not)y(not)+y(not)x+yx(not)d

Dz/yx	00	01	11	10
00	1	1	1	0
01	1	1	1	1
11	x	x	x	X
10	1	1	x	X

G=y(not)+d+x+z

We want to minimize the number of gates in the system. This is why we use this map. We have (or) gate and (and) gate and the (not).

A	B	Ab
0	0	0
0	1	0
1	0	0
1	1	1

This is the (and) gate. We can say this table is a (and) b.

We have also the (or) gate where we say (a + b) or (a or b). The logic gate table is given

B	A	a+b
0	0	0
0	1	1
1	0	1
1	1	1

We have also the (not) gate where the one gives zero and the zero gives one

Let us find the logic for x−or gate

Ba	00	01	11	10
	0	1	0	1

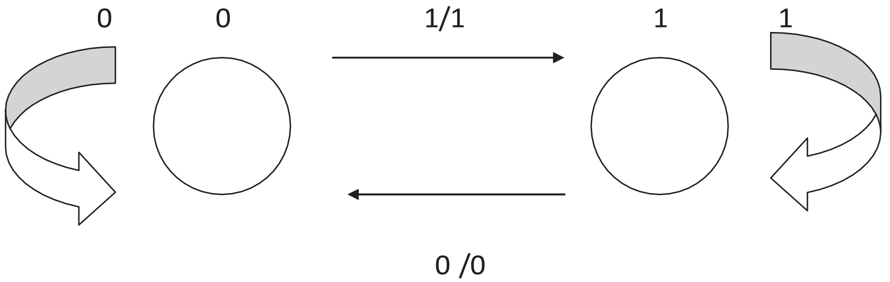
A xor b =b(not)a+a(not)b

All of this work is done to design the logic needed to take the four digit binary number dzyx to the seven segments of the seven segments display abcdefg.

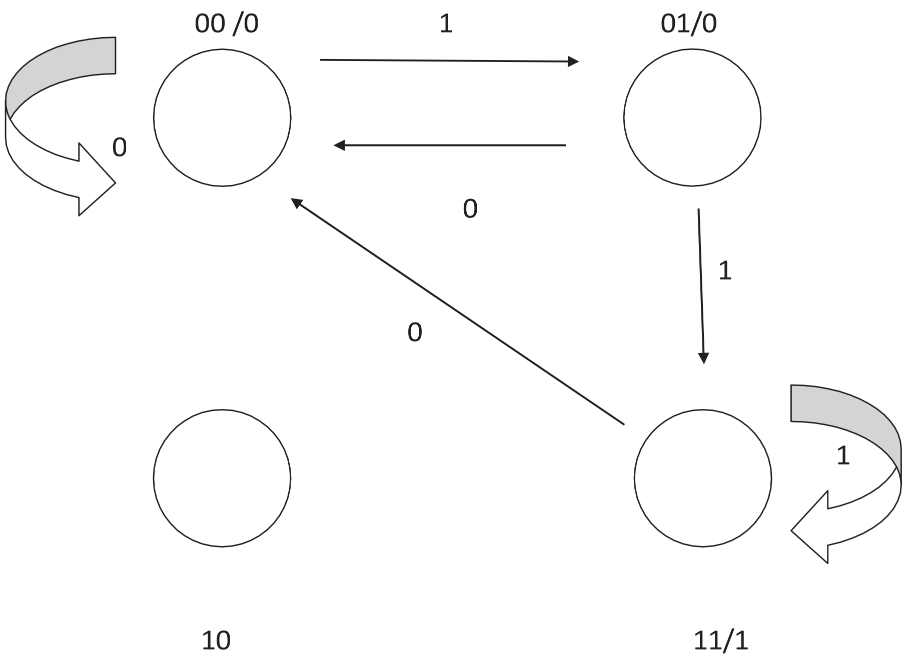
In doing this experiment we can take two of the segments just to show that they are (on) for the right number from 0 to 9. We can choose the easy logic to simulate like g and e.

To be practical we program an e−prom to the certain logic. The input is the four binary bits and the output is to the segments of the seven segments display.

Sequential Machines



State diagram



B	a	input	b+1	a+1	output
0	0	0	0	0	0
0	0	1	0	1	0
0	1	0	0	0	0
0	1	1	1	1	1
1	0	0	X	x	X
1	0	1	X	x	X
1	1	0	0	0	0
1	1	1	1	1	1

i/ba	00	01	11	10
0	0	1	1	0
1	X	X	1	0

A+1=i(not)a+ab

i/ba	00	01	11	10
0	0	0	1	0
1	X	X	1	0

B+1=ab

Output=b+1=ab

State	input	next state	output
0	0	0	0
0	1	1	1
1	0	0	0
1	1	1	1

Next state =input

Output = input

Sequential machines are a very good example of the use of the flip flop. We will study two kinds of sequential machines. The first is a function of the output and the state the second is a function of the state only.

First we have the state diagram. This diagram shows how we go from one state to the other according to the input. In this example we design a sequence detector that detect 11 and give an output of one when this is detected.

Then we move the data in the state diagram to tables to be ready to design the logic.

For the first machine we start at logic zero or state zero if the input is one we design the logic to give us one so that when we clock the d flip flop we go to state one.

We will talk about the d flip flop. The d flip flop has a state at the input when we clock it will be at the output. So it has an input and a clock and an output.

The number of d flip flops is the number of state bits. For our machine of one state bit and we are in state one if the input is one we design the logic as a function of the input and the state to have a next state one and an output of one.

The clock is the heart beat of the system as we can see the clock to the d flip flop is what makes the data moves in the system and going from one state to the other.

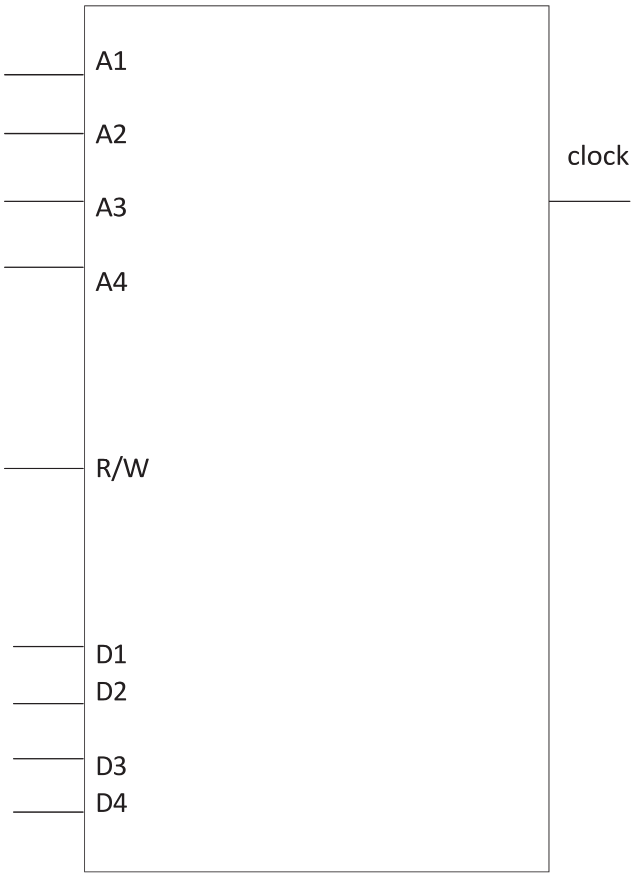
The state machine that the output is a function of the state only we start at state 00 and if the input is one we go to state 01. At this state if we have an input 0 we go back to the state 00 and if we have input of 1 we go to state 11 and the output will be 1 telling us that the sequence 11 is detected.

First we reset the d flip flops so we are at state 00. Then we design the logic if we have input one to give us 01 the second state to the input of the d flip flop. This state will be at the output of the flip flop when we clock it.

At this point we have a state 01 at the output of the flip flop. If the input is another 1 we design the logic to give us state 11 at the input of the flip flop when we clock it will be at the output. We do this according to the state diagram for all states and inputs.

Second we take the data to the tables and we make the map to design the logic. We use the map to minimize the logic needed to go from one state to the other. As we can see the clock takes the system from one state to the other according to the state and the input.

Read and Write to Ram



This Ram has four address bits so that the address can go from 0 to 15. It has four data bits to store numbers. In this experiment we are going to select an address and write to it data and then read what is in that address and check that it is the data we stored.

The address bus is in one direction and the data bus is in two directions.

The input to address or data can be 5 V for logic 1 and 0 V for logic 0.

Let us store numbers from 0 to 15 starting at address zero. First we select the address zero by putting 0000 to the address bus. We want to write data so we make sure that we select write. On the data bus we put 0000 and clock the system so that the first number is stored.

Second we select the second address and put 0001 to the address bus. We make sure that we want to write data by selecting W and we put 0001 to the data bus and clock it so that it is stored.

We keep doing this until the last address so that we have 0 to 15 stored in addresses 0 to 15.

Now it is time to read what is in the memory. First we select read. And we put the address that we want to read and let us say it is 0011 and clock the system. We will get 0011 on the data bus as output because it is the number that we stored.

This is a system to store in memory what we want and get it back when we want

Memory is needed in microprocessor systems and Ram is used to store results for example.

It is good to have an idea about the system architecture. Let us say we have a decoder. The input is the address four bits and it has 16 outputs that are going to select the four flip flops of the data.

The data is 4 X 16 D flip flops and the decoder selects four bits to read or write data. The D flip flop input and output is connected to a tri state buffer to the data bus.

If we want to write data on the data bus we clock it at the input of the d flip flop so that when we clock it is at the output. If we want to read data we connect the output of the flip flop to the data bus through the tri state buffer.

Hardware to multiply two numbers

In this work we will multiply two four bits numbers. The work is done using the hardware.

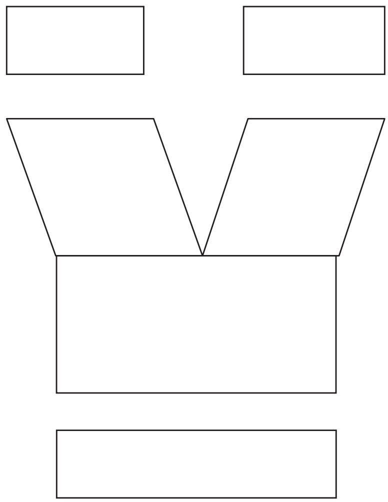
When we multiply two numbers in binary we start by the first bit of the second number. If it is one we copy the first number and if it is zero we copy zero. Then the second bit of the second number if it is zero we copy zero and if it is one we copy the first number shifted left by one and zero for the first bit. It is like when we multiply two numbers in decimal.

We keep going on until the last bit and at the end we add to get the results.

The system we are doing has an ALU and registers and shift registers and a control circuit. We have an input register for the first number and an input register for the second number and an output register for the result.

The ALU is used to add two registers and the result is stored in a shift register. It is also called the accumulator. The control sequential circuit is used to control the operation. Sometimes we have to shift and add and sometimes we have to shift only.

Let us say that we have two numbers of four bits in the two registers of eight bits. The first number is in the first register and the second is in the second register.



First we test the first bit of the second register if it is 0 we shift the first number one to the left and the first bit is zero and if it is one we add the first number to zero and shift to the left by one bit where the first bit is zero.

Second we test the second bit of the second number if it is zero we shift the first number by one where the first bit is zero and if it is one we add the first number shifted by one to the accumulator and put in the result in the accumulator. Then

Shift the first number by one to the left where the first bit is zero.

Third we test the third bit of the first number. If it is zero we shift the shifted first number by one to the left where the first bit is zero and if it is one we add the shifted first number by two to the accumulator and put in the accumulator then shift the first number by one where the first bit is zero.

We do this also for the last bit to get the result that will be in the accumulator.

As we can see to control this operation we need a sequential machine. First we draw the state diagram with all inputs and outputs and states.

Let us say that we make our design to complete the operation in 200 clock cycles. The first 50 cycles we test the first bit and operate accordingly. The second 50 cycles we test the second bit and operate accordingly and so on until we get the result.

As we can see we need an ALU and registers and the control unit which is the hardware program of the operation.

Electromagnetism (Electrostatic)

In this work we will talk about the parallel plat capacitor and the cylindrical capacitor.

First the parallel plats capacitor. The ideal case where we have uniform electric field between the two plats. Let us say that the upper plat at voltage 2.5 and the lower plate at voltage −2.5. In this work we will measure the voltage around and inside the capacitor.

By theory for the ideal case, if the field inside the capacitor is uniform we have

V=Ed

The equal potential surface is perpendicular to the field lines. This can be seen from the equation.

$$V=\int E.dx$$

By measuring the voltage inside the plates we will find it is uniform parallel to the plates and it is perpendicular to the field lines and the space between each surface and the other is uniform which mean that the electric field is constant and the voltage increase in a linear way.

The cylindrical capacitor

It has two cylinders the inner of radius (a) and the outer of radius (b). let us say that the inner cylinder is grounded and the outer at five volts.

The equal potential surfaces are circles around the inner cylinder and the field lines are radial perpendicular to them. To calculate the capacitance of the system we use

Q=CV

From gauss law

2 pi r E = q/€

$V=\int_a^b E.dr$

V=(q/2 pi €) ln (b/a)

C = 2 pi l € ln (a/b)

Static Magnetism

A loop of wire of current will generate magnetic field. We are interested in the field on the axis at the center of the loop and perpendicular to it.

This field is proportional to the current and is related to the radius of the loop and the distance to the center of the loop. The further we are from the loop the lower the magnetic field.

It will be interesting to look at two loops where the centers of the loops are on the same axis. The two loops are separated by a distance d.

The axis perpendicular to the loops which pass by the centers is of interest to us. The field on this axis will start high at the first center then will drop and will be the lowest at the middle of the two centers.

If the current was the same in the two loops the curve will be symmetric around the middle

In this work we will find the field equation for the single loop and for the two loops.

$H_z = I R^2/2(R^2 + z^2)^{3/2}$

This is the H field for one loop, where R is the radius and z is the perpendicular distance to the center of the loop.

If we have two loops separated by a distance d, we want to find d where not only the first derivative is equal to zero but also the second.

This curve starts at zero concave down. As z increases it should be concave up at some point because it does not cross the axis. When the curve change from (concave down) to (concave up) the second derivative will be zero.

$H_z = I R^2/2(R^2 + z^2)^{3/2}$

$dH_z/dz = I R^2/2(R^2 + z^2)^{5/2} * 2z$

$d2H_z / dz^2 =$

$5/2 I R^2/2(R^2 + z^2)^{7/2} * 2z + I R^2/2(R^2 + z^2)^{5/2} * 2$

$= 0$ This is true for z=R/2

Transformers

The magnetization current in a transformer is not pure sin wave. In this work we will find the ratio of the strong harmonics of this current.

First we will generate the wave that we need to study. Second we will project it on its harmonics to find the ratio of these harmonics. We will do projection in a digital way.

The original wave that we want to study is sampled at 20 samples for one full wave.

If we want to test the third harmonic component we first generate this third harmonic and we also sample it to 20 samples. We then multiply each sample of the wave by the corresponding sample of the third harmonic. Then we sum all samples and dived by 20.

We do this for the fundamental and third harmonic. We will then find that there is a strong third harmonic component and this is due to the hysteresis loop that we have.

f(1)		sin(2pi/20)		sin(2pi3/20)		
f(2)		sin(4pi/20)		sin(4pi3/20)		
f(3)		sin(6pi/20)		sin(6pi3/20)		
f(4)		sin(8pi/20)		sin(8pi3/20)		
f(5)		sin(10pi/20)		sin(10pi3/20)		
f(6)		sin(12pi/20)		sin(12pi3/20)		
f(7)		sin(14pi/20)		sin(14pi3/20)		
f(8)		sin(16pi/20)		sin(16pi3/20)		
f(9)		sin(18pi/20)		sin(18pi3/20)		
f(10)		sin(20pi/20)		sin(20pi3/20)		
f(11)		sin(22pi/20)		sin(22pi3/20)		
f(12)		sin(24pi/20)		sin(24pi3/20)		
f(13)		sin(26pi/20)		sin(26pi3/20)		
f(14)		sin(28pi/20)		sin(28pi3/20)		
f(15)		sin(30pi/20)		sin(30pi3/20)		
f(16)		sin(32pi/20)		sin(32pi3/20)		
f(17)		sin(34pi/20)		sin(34pi3/20)		
f(18)		sin(36pi/20)		sin(36pi3/20)		
f(19)		sin(38pi/20)		sin(38pi3/20)		
f(20)		sin(40pi/20)		sin(40pi3/20)		

At this point we multiply each sample of the function by the third harmonic and then divide by the number of samples.

We do this for the first harmonic and the third harmonic and then we add the result for the third harmonic to get the ratio of harmonics.

By using this method we will find that there is a strong third harmonic in the signal.

Induction Motor

In this work we will generate the torque versus speed curve and plot it on a plotter.

The motor we have in the lab has a sensor of speed and this is one axis. If we take the derivative of speed we will get something proportional to the torque and this is the other axis. So we have a torque axis and a speed axis.

To make the differentiator we can use a differentiator circuit using amplifier or try to estimate it using RC network. The transfer function for an RC circuit is

$$V_{out}(s)/ V_{in}(s)=Cs/1+RCs$$

When $RCs << 1$ this will operate like a differentiator and the transfer function will be Cs approximately.

This is an RC circuit where the input is v in and the output is from the resistance R v out

Measurement systems

The needle meter is a sensitive device for current. It has a full scale current of 100 micro Ampere. It can be used to measure current or voltage.

First we are going to measure current. To measure current we are going to connect it in series with the branch. The meter has a small resistance let us say 100 Ohm. We are going to do our design such that the current we have is a percentage of the full scale. To do this a resistance connected to the meter in parallel is needed.

Let us say we want to measure 1 m A full scale current. The full scale needle current is 100 micro A. The resistance is 100 Ohm which make the current .01 m A.

We use a resistance in parallel to the needle scale so that it has the 1 m A for full scale. .01V/.001A is ten Ohm for the resistance in parallel.

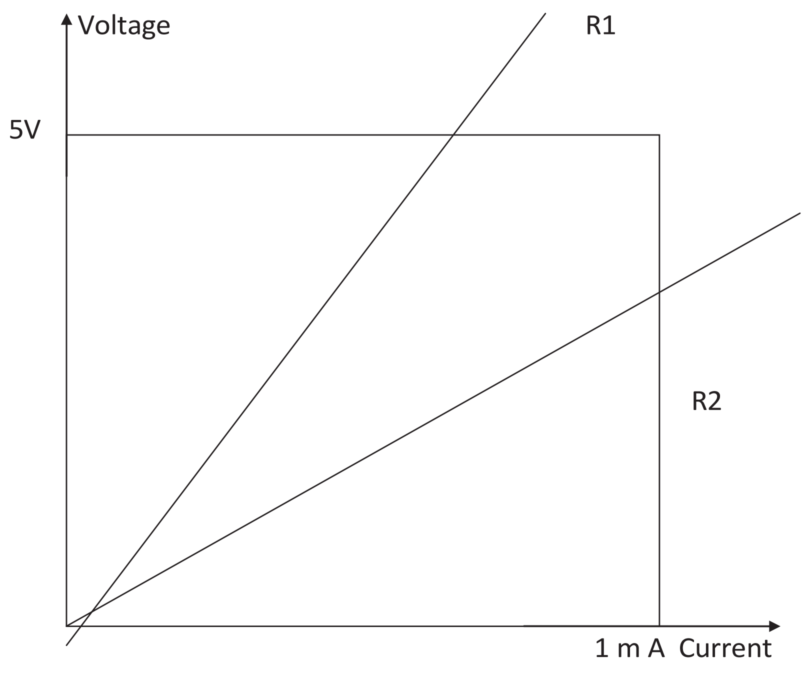
When we have two resistances in parallel the equivalent resistance is less than the smallest resistance. Because we have small resistance for the needle scale it will be smaller with a resistance in parallel.

The Voltmeter

An ideal voltmeter has an infinite internal resistance. The voltmeter is connected in parallel to the branch or resistance we want to calculate the voltage at.

Let us say we want to measure five volts with the needle meter. And its current is 100 micro A. The resistance of the voltmeter has to be 50000 Ohm. This resistance is the sum of 100 and 49900.

The power supply



An ideal power supply gives a constant voltage for any current. This is not possible in practice because power is limited. If we have a resistance R1 as we increase (V) the (I) will increase in a linear manner where the slope is R1.

If R were large the power supply can deliver the current for the 5 V let us say.

In real power supply we have a current limiter where we can set the max power to be delivered. It is the max current delivered by the power supply.

Let us start with a large resistance so that the (operating point) at the 5 volt curve. So far the power supply is operating at the normal region.

Let us say we decrease the resistance until the current is 1 m A. This operating point is for max power from the power supply.

If we decrease the resistance more the current will stay the same and the voltage will drop. The operating point will be on the 1 m A curve.

The oscilloscope

In the oscilloscope we have an electron gun that shoots the electron beam to the screen. In the way of the beam there are vertical and horizontal deflection plates.

On the horizontal deflection plates there is a time based signal. This signal moves the dot in the screen from left to right with a certain speed.

On the vertical deflection plates we apply the signal that we want to see as a function of time.

The screen is divided to 20 vertical divisions and ten horizontal. We have two switches to adjust the scale of the vertical and horizontal divisions.

The x−axis is time and adjusting the switch is choosing the right time based signal.

The y−axis is the voltage of the signal that we want to measure and adjusting the switch is making the right V / Div for our signal.

The time based signal is a voltage as a function of time. The voltage start at zero and increases in a linear manner until the dot is at the right on the screen.

This mean that when the voltage is zero the dot is at left of the screen. The time based signal is applied to the horizontal deflection plates.

When the dot reaches the end of the screen to the right the time base signal goes back to zero.

As we know the deflection is parabolic as a function of voltage and we need it to be linear with voltage. We have to take this into consideration.

To include this fact in our design we have to adjust the parallel plates in a way so that the deflection is linear with voltage.

This is one way. Another way is to have a circuit that takes the square root before we input the voltage to the parallel plates. This way the deflection will be linear with the input voltage that we want to see or use to scan.

After the time based signal goes back to zero it stays at zero until the input signal reaches a fixed certain positive values that will trigger to repeat the time base signal.

This way we will have a point reference of the input signal that we will start from.

It is important to synchronize the reference point to start the input signal and to start the time based signal. This way we will keep the image fixed on the screen.

The dot starts at the left and if the time based signal is slow we can see it moving slowly to the right.

As we increase the time based the dot moves faster until our eyes cannot resolve the dot and we will see a line.

The time based signal where we start not being able to resolve the dot by our eyes is important for biomedical engineers.

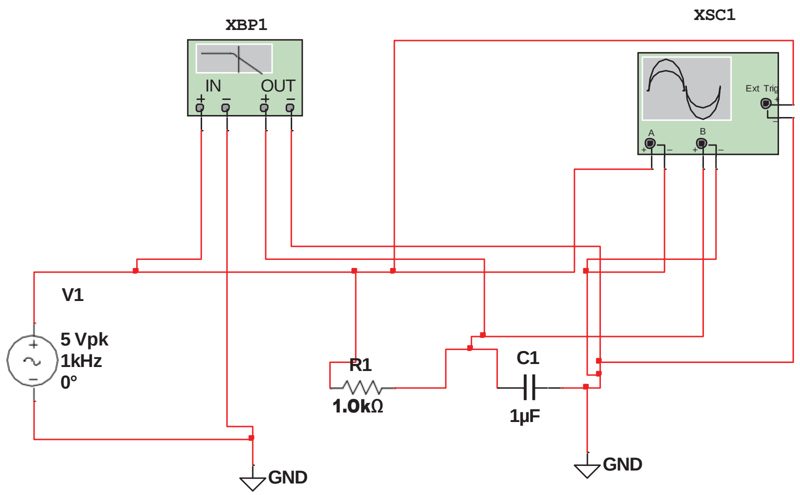
In the scope we have AC mode and DC mode. The AC mode measure AC and low pass filter any DC component. The DC mode measure AC plus DC.

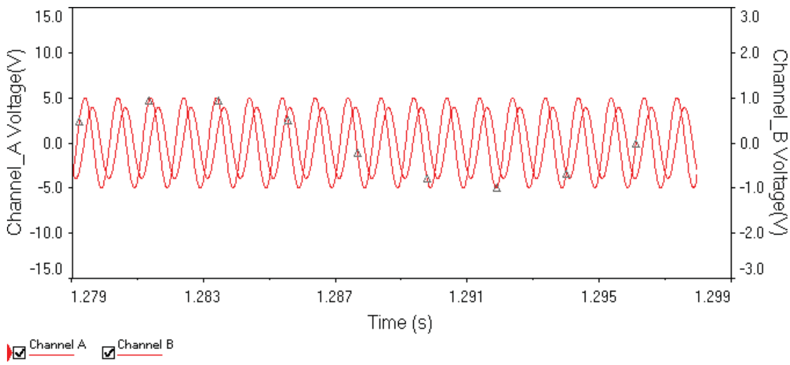
Applications of the oscilloscope

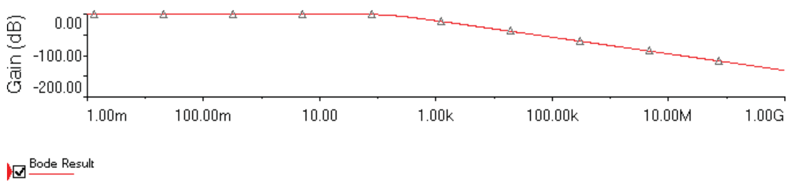
We can use the oscilloscope to measure the gain of a sin wave and the phase shift between two sin waves.

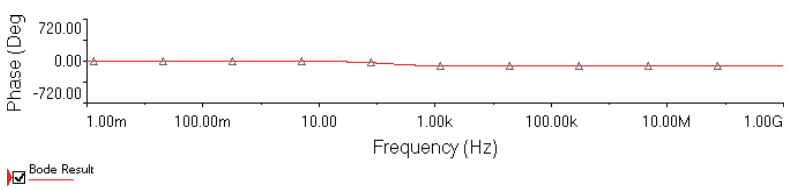
To do this we will use a simple RC low pass filter. The input goes from the function generator to (RC) circuit and the output taken from (C).

The input goes to channel one and the output goes to channel two. We can measure the output as a function of the frequency and the phase shift




15.0
10.0
5.0
0.0
-5.0
-10.0
-15.0
Channel_A Voltage(V)
3.0
2.0
1.0
0.0
-1.0
-2.0
-3.0
Channel_B Voltage(V)
1.279
1.283
1.287
1.291
1.295
1.299
Time (s)
Channel A
Channel B


Gain (dB)
0.00
-100.00
-200.00
1.00m
100.00m
10.00
1.00k
100.00k
10.00M
1.00G
Bode Result


Phase (Deg
720.00
0.00
-720.00
1.00m
100.00m
10.00
1.00k
100.00k
10.00M
1.00G
Frequency (Hz)
Bode Result

We can see the input sin wave and the output sin wave with a phase shift and a gain. We can measure the output and measure the phase shift between the input and output.

We can do that for many frequencies and see how the circuit will operate like a low pass filter.

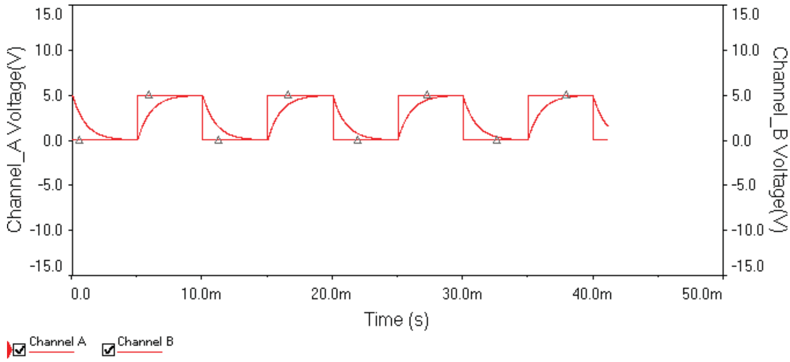
Also we can measure the phase shift for every frequency. We will see that for the low pass region the phase shift is zero and for the half power point the phase shift is −45 degrees and for very high frequencies the phase shift is −90 degrees.

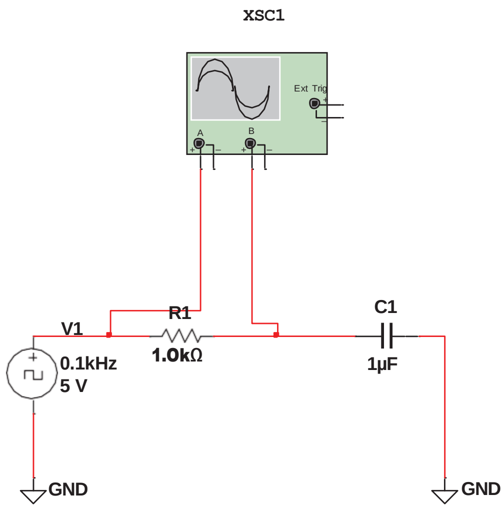
This is a low pass filter where the half power point is 1/RC. If we take the output from R it will be a high pass filter.

For the high pass filter if the frequency is << 1/RC the output is very small and the phase shift zero.

At the half power point where the frequency is equal 1/RC the phase shift is −45 degrees.

At frequencies >> 1/RC the output is high and the phase shift is approximately 90 degrees.





For this circuit we have a first order differential equation where the derivative of the function is equal to the function. The solution to it is an exponential where we have to satisfy initial conditions.

The equation in the time domain is [15]

Q=CV for the capacitor

I= C dv/dt taking the derivative

Vin – I R – 1/C ∫ I dt =0 for the loop

We want to get the step response. This can be done by going to the frequency domain where (V in) in the frequency domain is 1/s and we solve for V out.

1/s− I R −I / Cs =0

C −I R C s − I =0

I = C /(RCs+1)

V out =1/s(RCs+1)=1/s − RC / (RCs+1)

If we go back to the time domain this is a step minus a damped exponential which is what we see on the simulation graph.

The solution is

Vout(t) =(A1 + A2 $e^{-t/RC}$) u(t)

A quantity that is important to us is the time constant RC. The damped exponential function drops by 77% after one time constant.

After five time constant less than 2 % is left of the damped exponential. This is important because this time constant and the gain completely describe the function.

What about the high pass filter or if the output is taken from the resistance? We have the same loop equation and the same current and the output will be

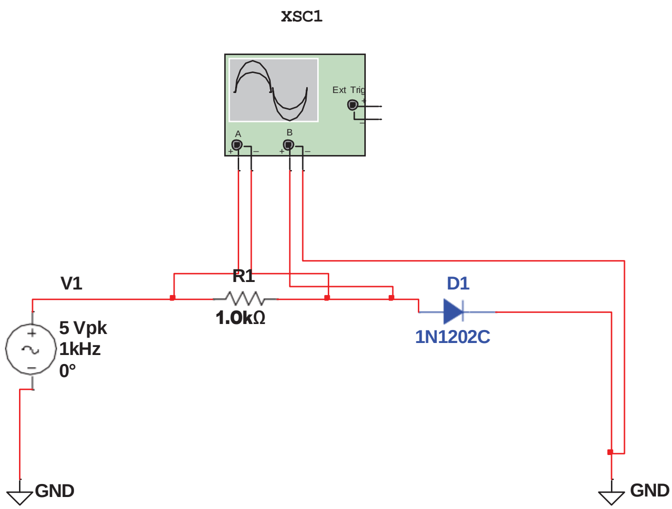
VoutR(s)= RC / (RCs+1)

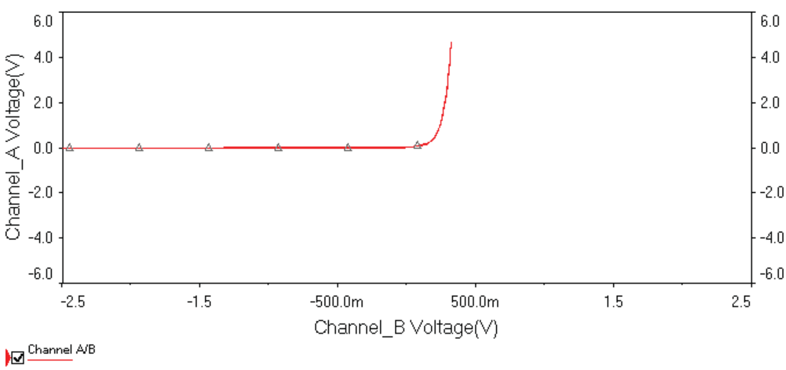
To get the solution we have to go back to the time domain and when we do that we get

Voutr(t) = $e^{-t/RC}$u(t)

The transfer characteristic for the diode

Another application of the oscilloscope is to get the transfer characteristic of the diode. The scope in the x–y mode can show the transfer characteristic where we have one channel as the x–axis and the other as the y–axis.





We want to get I versus V curve. To do that we take for channel x the voltage of the diode and we take for channel y the voltage of the resistance which is proportional to the current.

As we can see this is not an ideal diode where in the ideal diode the transfer characteristic is the negative x−axis and the positive y−axis.

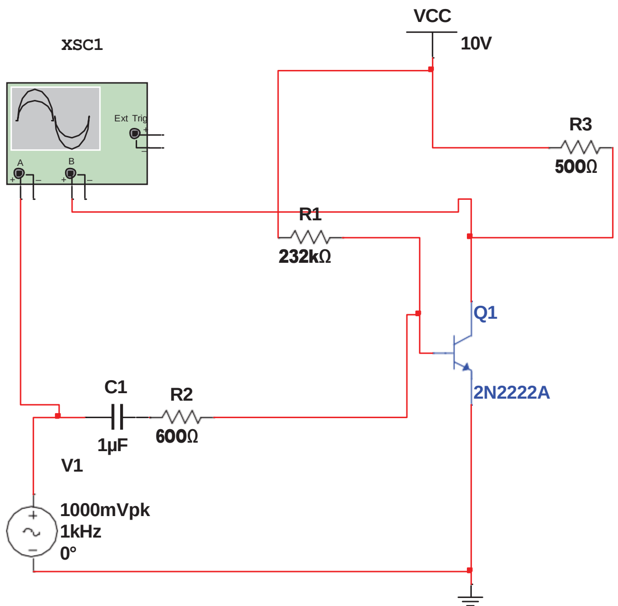
What can be a good model for the diode? First we have to note that the diode works on two states. For an ideal diode any positive voltage the diode is a short circuit and any negative voltage the diode is an open circuit.

A more real model is a (two states) where the characteristic of the ideal diode is shifted by .7 V to the right. In this model when the diode is short we have to add a battery of .7 V from positive to negative in the direction of the diode.

Another model is a two state model where on the state where the diode is short we have a battery and a small resistance. This model better estimates the real diode.

We do this to get a linear system that we can solve using linear circuit theory.

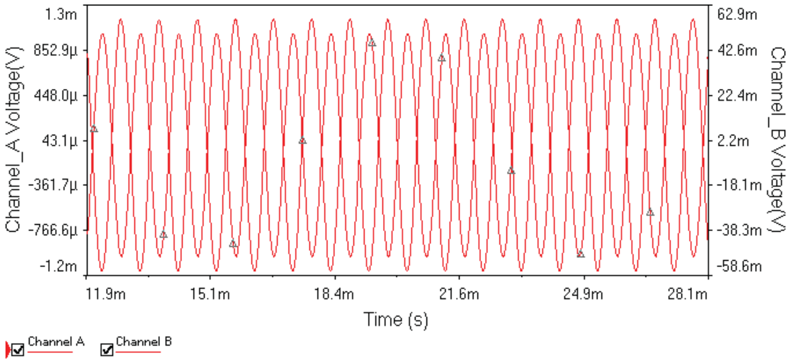
Electronics One



One thing that a transistor is used for is analog amplifier. First we have to dc bias the transistor to get to the operating point. In our design we use a capacitor to block the bias from the input. In our design we have to make sure that the capacitor is a short circuit seen by the input. By that I mean its impedance 1/jwC is small comparing to the resistive input.

First the bias and to get the base current we have the loop

$10 - I_b\,(232) - .7 = 0$



We can see that Ib is equal to 40 micro A. for this (Ib) the (Ic) is equal to 8 m A.

B=200 for this transistor

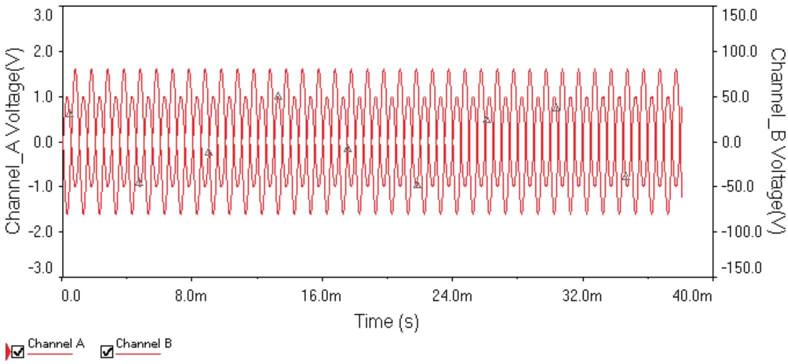
g= Ic/Vt =8/25=.32

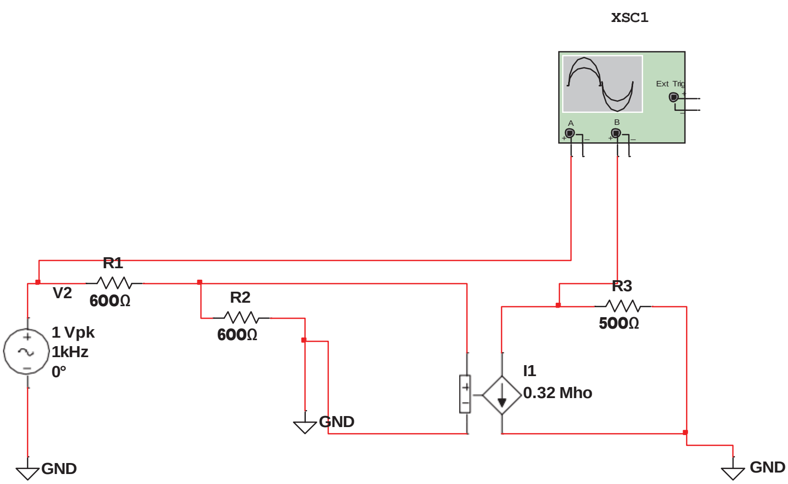
rπ=β/g=625 Ohm

To calculate the gain we have to draw the small signal model.

V in /2 *.32 *500=v out

V out / v in = 80

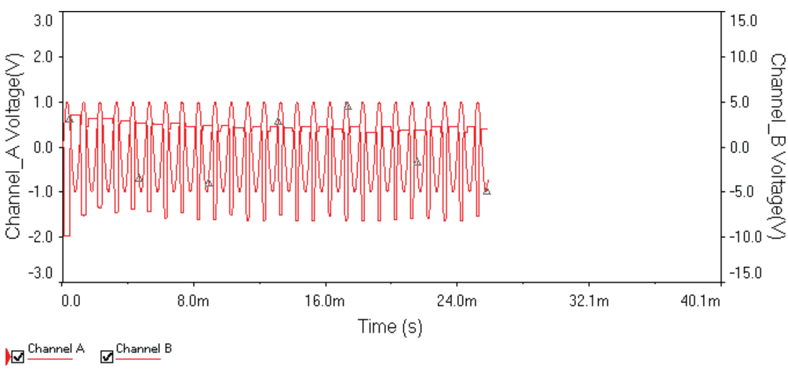




This is the ac analysis. After we get to the bias point we calculate all the needed parameters and forget about the dc. From this point we draw the ac model with the needed parameters and start calculating the gain.

For the transistor we have dc to get to the bias plus the ac that we want to amplify. First we bias the transistor then we get $r\pi$ and g from Ic of the bias.

The dc response is dc and the ac response is ac. The superposition of the two is the complete response. To simplify things we first do the dc analyses and get to the operating point. At this point we get all the parameters needed and forget about the dc and do the ac analyses.



One thing we have to know that the transistor does not generate power but it control power. By that I mean that a small amount of energy can be used to control a large amount.

This is why we use the transistor as an amplifier. We have a small signal that cannot drive the load we use the transistor to amplify so it can be used.

Two limitation of the transistor are saturation and cutoff. We will study each one carefully.

The first limitation is saturation. The reel supply and the resistance have a load line. The transistor has a characteristic curve that meet the load line at the operating point. The operating point changes as a function of the input base current. And for every base current we have a characteristic curve.

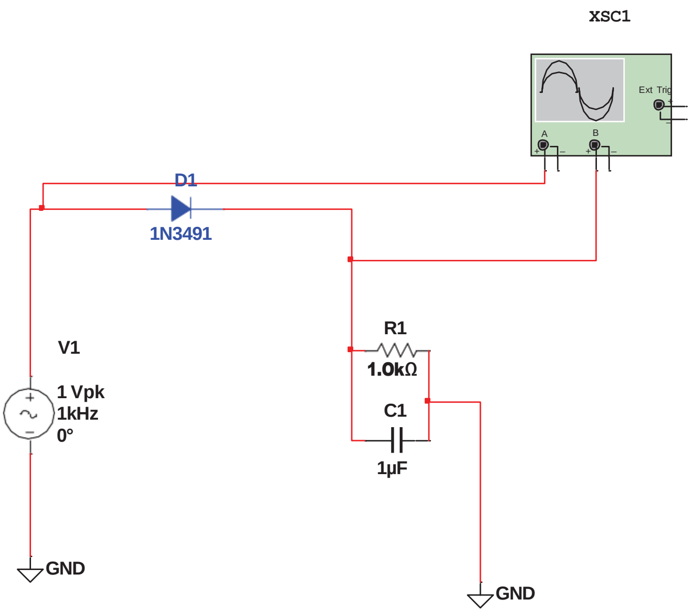
If we inject more and more current into the base the characteristic curve will rise and when it meets the load line at .2 volts we will have saturation.

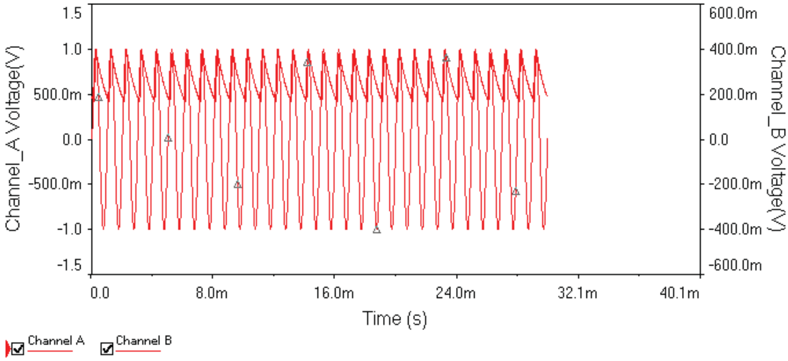
The second limitation is cutoff. One can feel cutoff when we have no input to the transistor so that the reel voltage is across the emitter and collector of the transistor.

When we are at the operating point and we decrease the base current the collector current will decrease and the output voltage will increase until the collector current is zero so we have cutoff.

We can see the two limitations in the graph. One has to design where the operating point and the load line such that we do not fall into these limitations. We have to have a bound for the input so that we avoid falling into these limitations.

Power supply





In this work we want to convert AC to DC. To do this we will use a resistance and a capacitor and a diode.

When we have the positive cycle the diode is short and the capacitor will charge until we get to the top of the cycle.

At this point the AC will decrease and the voltage across the capacitor will see an open circuit in the diode path and the output voltage will come down in an exponential decay as if it is an RC circuit.

If the time constant was large comparing to the period of the sine wave we will have good dc. If the time constant was small comparing to the period of the sine wave the dc will have distortion.

We can calculate the boarders of the fluctuation of the dc by knowing the period and the time constant. For example if we allow 2 % decrease of the dc every period of the sine wave we have to shows R and C so that the time constant is large enough comparing to the period to meet the design specifications.

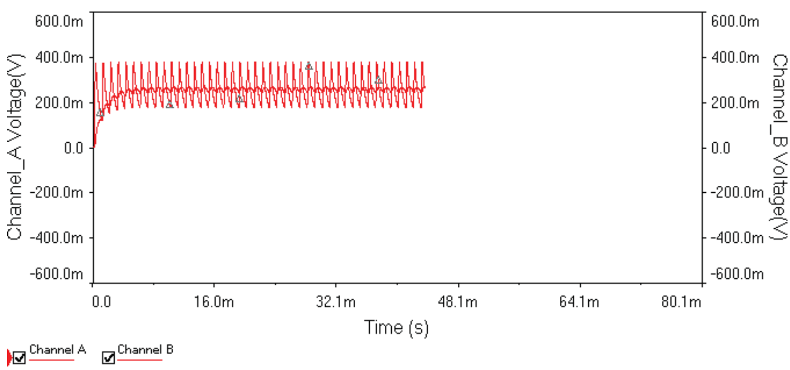
We can derive an approximate formula that relates the percentage allowance of fluctuation and the choice of R and C of the circuit.

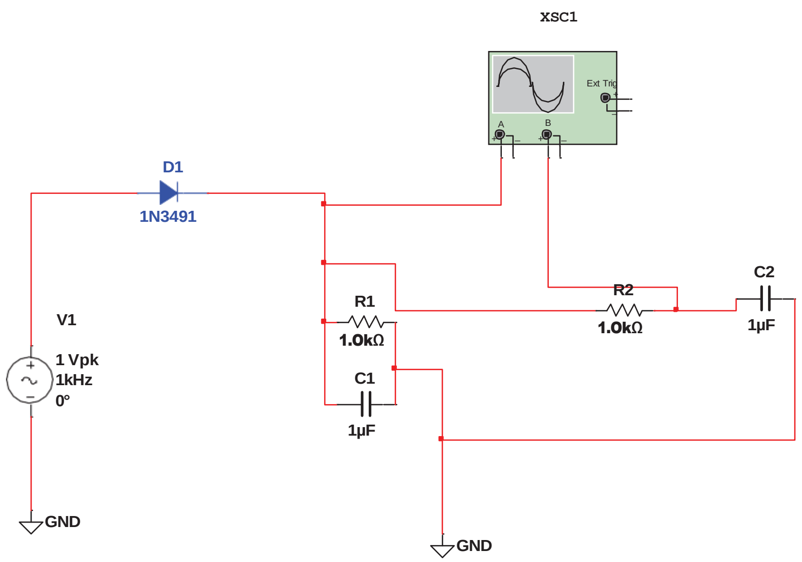
$e^{T/RC}$ = minimum constant in %

We made the approximation that the min is after time T which is the period of the sin wave.

For a minimum constant in % let us say 95 % which allow 5 % error from the max we can find R and C because T is known.

As we can see in this circuit we converted ac to dc and we can design for certain percentage error.





To make the output smooth we connect it to a low pass filter. This will make the dc more pure and will remove the oscillation.

The graph shows this in a clear way. The low pass filter will pass only the dc.

Fourier transform of time limited digital sinusoidal

In signals and system we have time domain and frequency domain. We go from the time domain to the frequency domain to solve the system.

An example to illustrate this is if we have two numbers and we want to multiply them. We can take the log of each number then add the two logs then take the antilog of the result.

A sin wave can be described as continues wave of time or as an impulse at a certain frequency on the frequency axis.

For analog signal to go from (time domain) to the frequency domain we take the Fourier transform and for a digital signal we take the z−transform.

What we have in practice is a time limited sinusoidal. This a sin wave in a rectangular window from zero time to T. and in this work we are interested in a digital signal.

To go from an analog signal to a digital one we need ADC which will sample the analog signal every Ts.

The signal is [14]

$x(t)=a \sin(wt + \varphi)$

Sampling

$x(n\,Ts)= a \sin(2\pi (f/fs)\, n + \varphi)$ where $fs = 1/Ts$

This is a digital signal. Let us say that n=1,2,3… ….,N

This signal in the time domain and we want to find the Fourier transform.

$$X(jw) = a \sum_{n=0}^{n=N} e^{j2\mathrm{pi}fn}\ a/2\ \left(e^{j(2\mathrm{pi}Fn+\ \varphi)} - e^{-j(2\mathrm{pi}Fn-\ \varphi)}\right)$$

Let us ignore the second term for the sin wave because it will be the mirror image

$$X(j2\pi f) = a \sum_{n=0}^{n=N} e^{j2\mathrm{pi}(f-F)n-j\ \varphi}$$

This is for one sin wave and if we have two we can use superposition

Using the fact that

$\sum_{n=0}^{N}$ a^{n}= (1− a^{N})/ (1 − a)

X (j2πf) = a e $^{\varphi\,j}$ (1 – e$^{j2pi(f-F)N}$) / (1 – e$^{j2pi(f-F)}$)

Taking half the exponential out

X (j2uf) = a e$^{-j\,\varphi}$ e$^{j2pi(f-F)(N-1)/2}$(sin (2π(f−F)N)/ sin(2u(f−F))

This is for one sin wave. For the description to be complete we need magnitude and phase.

If we have two sin waves of F1 and F2 by looking at the frequency response we can resolve the frequency.

As F1 and F2 get closer for some φ we will not be able to resolve the two frequencies. We need more window or more samples (N) to do that.

This is important because we can find a formula for the limit on the delta F that we can resolve.

Let us say the first digital frequency is .2 and the second is .25. For $\varphi=0$ we cannot resolve the two frequencies and to be able to resolve them we need more window or more samples (N).

For the same digital frequencies and $\varphi=\pi/2$ we are still able to resolve.

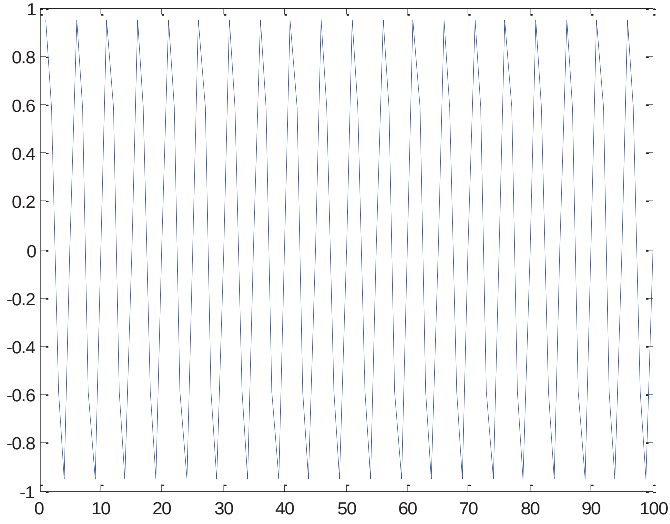
For the same digital frequencies and $\varphi=\pi$ we are able to see two peaks and resolve.

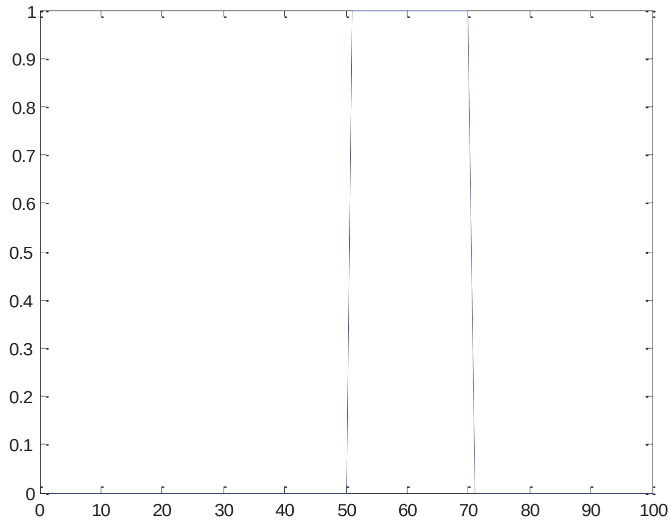
Using superposition when we add in the time domain we also add in the frequency domain.

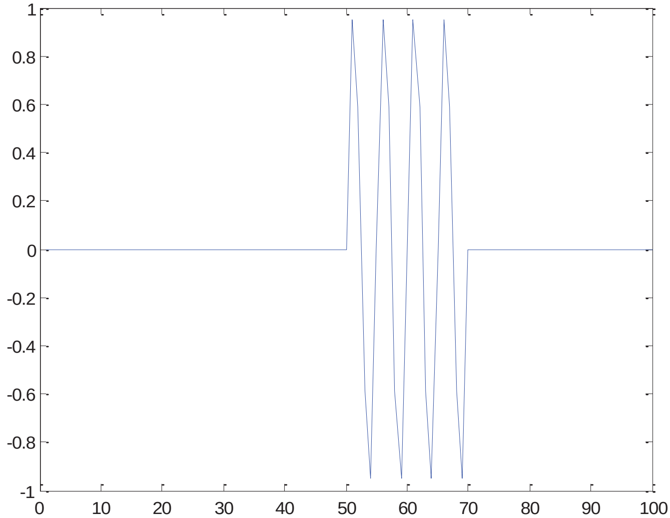
The single sin wave has the spectrum that we derived and the sum of two sin waves has the sum of the two spectrums for each one.

If F1 and F2 was close let us say .2 and .25 we do not add magnitude quantity but phase and magnitude. For example the point in the middle of F1 and F2 has the same magnitude of spectrum due to the first and second sin wave. But the relative phase changes with φ.

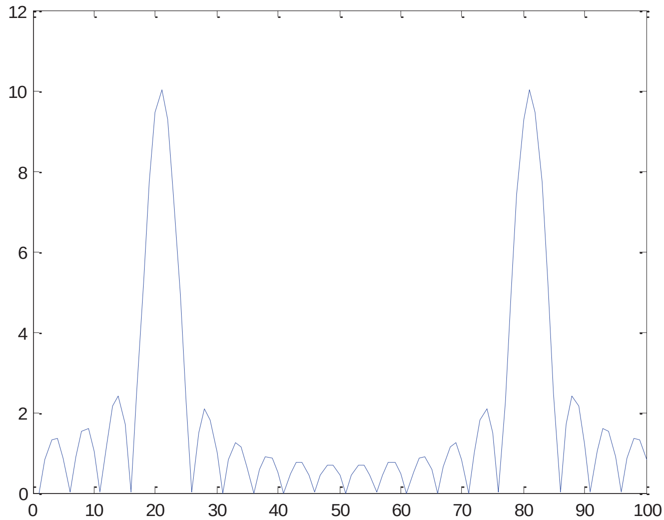
For our example when $\varphi=0$ we have two in phase quantities and they add to a single peak in the middle.

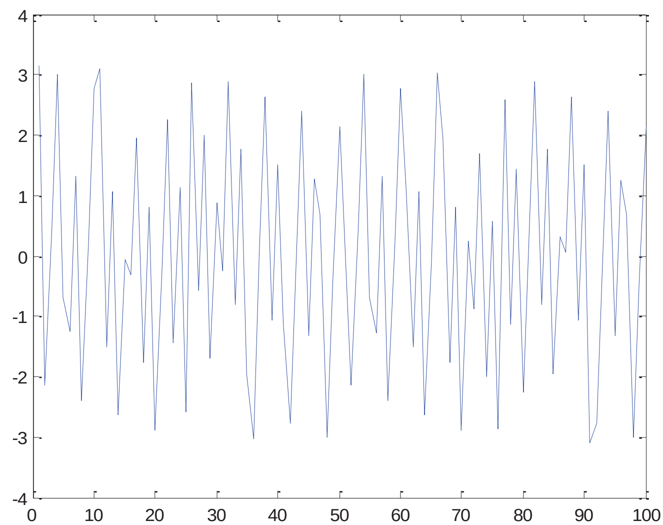

1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0
10
20
30
40
50
60
70
80
90
100


1
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
0 10 20 30 40 50 60 70 80 90 100


1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0 10 20 30 40 50 60 70 80 90 100

x(t)=a sin(wt + ф)

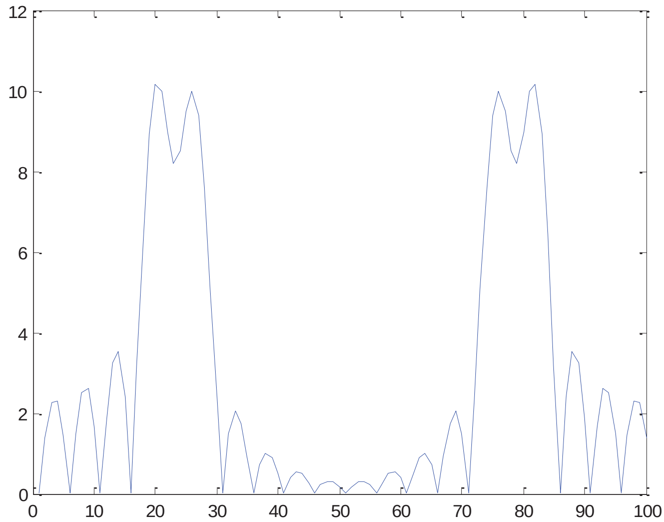

0 2 4 6 8 10 12
0 10 20 30 40 50 60 70 80 90 100


-4 -3 -2 -1 0 1 2 3 4
0 10 20 30 40 50 60 70 80 90 100

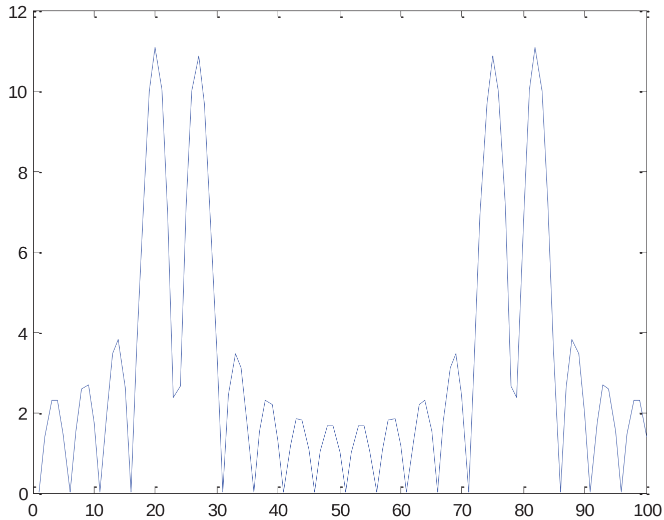
phase

Matlab program

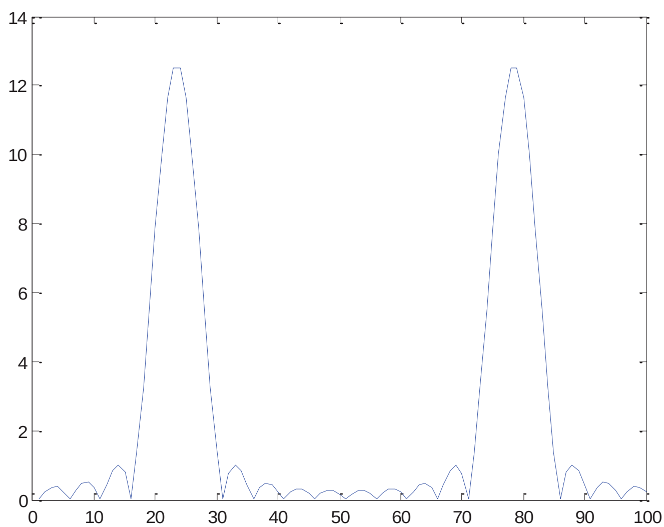
```
t=1:1:100
for t=1:50
    s(t)=0
end

for t=51:70


    s(t)=1
end

for t =71:100


    s(t)=0

end

plot (s)

t=1:1:100

x=sin(2*pi*.2*t)

plot (x)

w=x'*s


for t=1:100


    d(t)=w(t,t)

end

plot (d)

q=fft (d)

m=abs (q)

plot (m)

n=angle (q)

plot (n)
```



$x(t)=a \sin(w1t) + a \sin(w2t + \pi/2)$



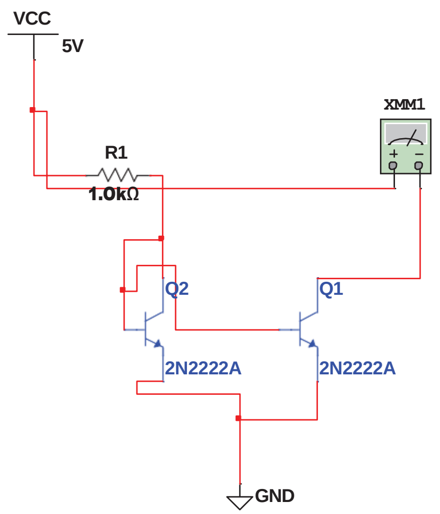
x(t)=a sin(w1t) +a sin(w2t + u)



x(t)=a sin(w1t) +a sin(w2t)

Electronics two

Current mirror



As we can see one transistor is a diode. The current in the 1 K ohm resistance is

(Vcc −vd)/R =(5 − .6)/1 =4.4 mA

As we can see Vbc of the first transistor is the same as the second transistor. As a result we will have the same base current in the two transistors.

By that we can see that we can shows R for the needed current and the same current will be in the mirror side no matter what the load is .

Vbc is .6 volts. (Vcc −Vbc)/R is the current that we want. Ib1=Ib2 because Vbc is the same as a result we will have a current source.

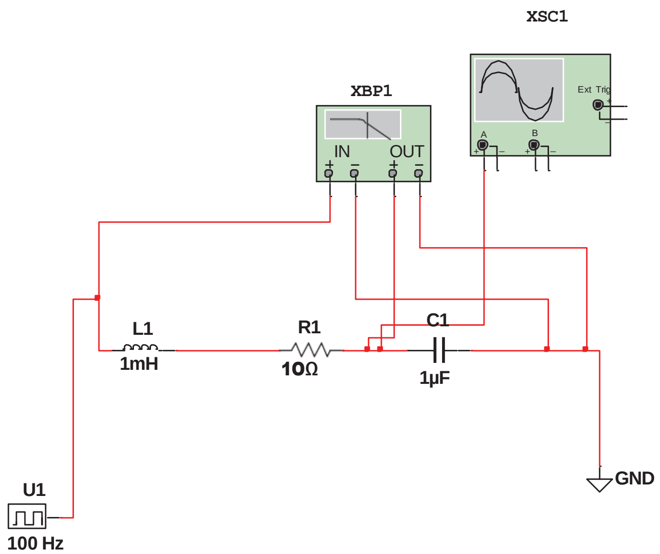
Two cases we have to consider for the operation of this current source. One is saturation and the other is cut off. We will talk about each one.

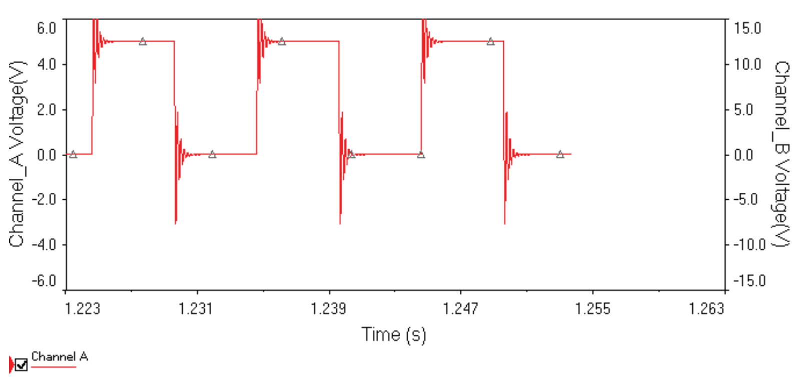
If we have a current in the base we will have a fixed characteristic curve. The load line of Vcc and R load is a straight line.

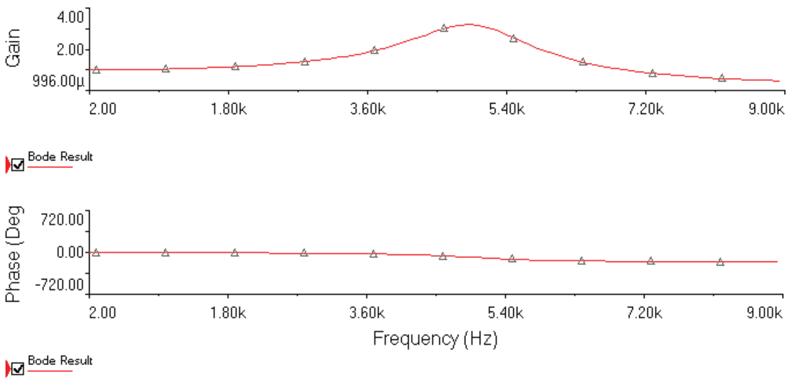
If we have a voltage V and load R the maximum current can be

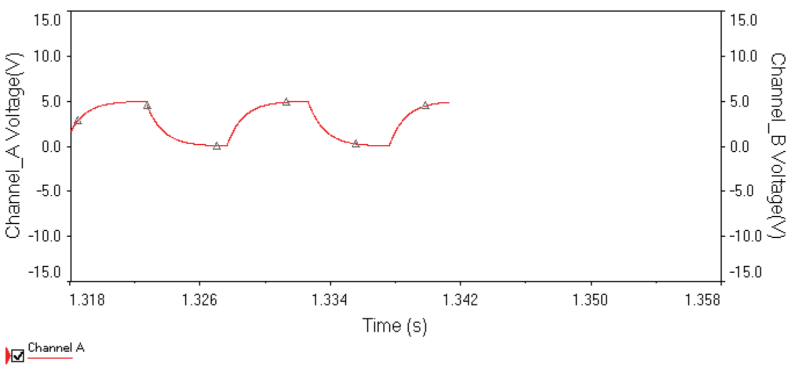
V/R. if Icc is less than this value we will have I cc as an output of the current source. If Icc is greater than V/R we will have saturation.

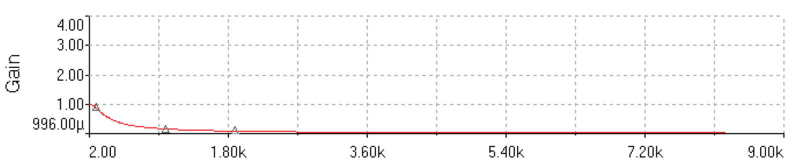
Tuned circuit






Gain
4.00
2.00
996.00μ
2.00
1.80k
3.60k
5.40k
7.20k
9.00k
Bode Result
Phase (Deg
720.00
0.00
-720.00
2.00
1.80k
3.60k
5.40k
7.20k
9.00k
Frequency (Hz)
Bode Result


Channel_A Voltage(V)
15.0
10.0
5.0
0.0
-5.0
-10.0
-15.0
Channel_B Voltage(V)
1.318
1.326
1.334
1.342
1.350
1.358
Time (s)
Channel A


Gain
4.00
3.00
2.00
1.00
996.00μ
2.00
1.80k
3.60k
5.40k
7.20k
9.00k

We are talking about three elements in series. First a resistant second a capacitor third the inductor.

For the resistance we have Ohm law

$V=R\ I$

For the capacitor we have

$Q=C\ V$ and by taking the derivative

$I = C\ dv/dt$

And for the inductor we have

$\Phi = L\ I$ and by taking the derivative

$V=L\ di/dt$

The three elements are in series and if we write KCL for the loop

We have

$$V_{in} = I R + L\, di/dt + 1/C \int i\, dt$$

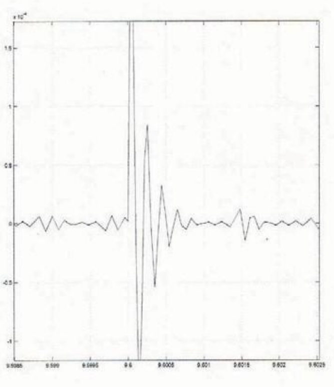
Taking the derivative we have

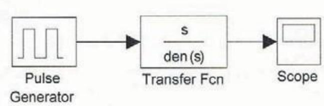
$$dV/dt = R\, di/dt + L\, d^2I/dt^2 + 1/C\, i$$

Taking the Laplace transform with zero initial conditions

$$s V(s) = R\, s I(s) + L\, s^2 I(s) + 1/C\, I(s)$$

$$I(s)/V(s) = s/(s^2+(R/L)s+(1/(CL)))$$

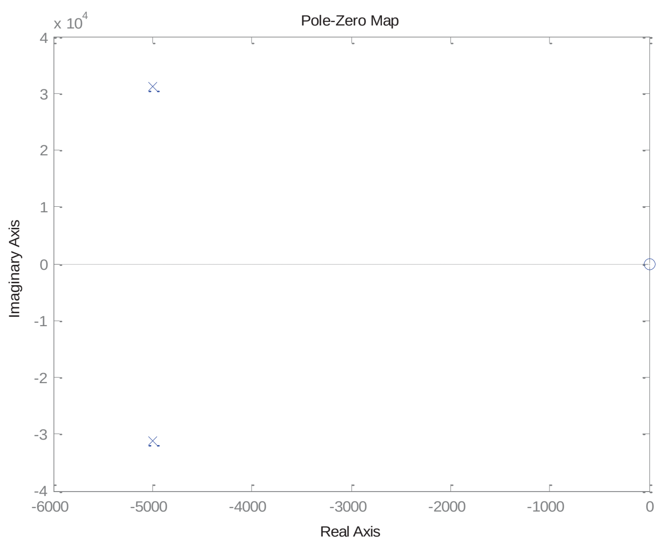


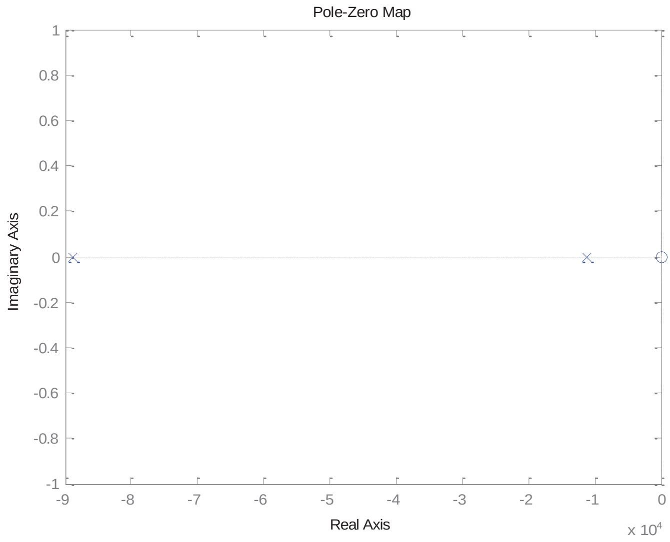


This transfer function for the current or proportional for the voltage in the resistance. For this transfer function we have two poles and one zero.

The poles have to be in the left side of the plane for the system to be stable and the negative real part give an exponential decay and the imaginary part give the natural frequency of the decay.

The parameters of the system determine if the response will oscillate or not. For example in our system if R=1000 ohm we will not have an imaginary part of the poles and they are negative and real and the response for the step input will not oscillate.





The system has two poles

$$P1= (-b +\sqrt{b^2 - 4ac}\ )/2a$$

$$P2= (-b -\sqrt{b^2 - 4ac}\ )/2a$$

Using our system

$$P1= -(R/L) +((R/L)^2 - 4/LC\ )^{1/2}/2$$

$$P2= -(R/L) -((\ R/L)^2 - 4/LC\ )^{1/2}/2$$

As you can see for our L and C if R=1000 ohm we will have two negative real poles. As R gets smaller the two poles will get closer on the real axis until they meet. At this instant we say the system is critically damped [13].

From this point as R get smaller we will have poles that are complex conjugate to each other and the response will oscillate. We have an example for R=1000 ohm and R=10 ohm. We have the poles and the step response.

Another way to look at this LTI system is the frequency domain. We know from LTI system theory that if the input is sinusoidal the steady state output is sin wave with a gain and a phase shift for every frequency.

This can be seen from the differential equation of the system by replacing the input voltage by sinusoidal and solving for the current.

To make things easier we take Vin= A cos(wt) and use the system equation to find the current.

The way I will do it is a good introduction to phase approach. We will see that for this input all outputs in steady state have the same frequency.

$A w \cos(wt+90) = = L\ d^2I / dt^2 + R\ di / dt + + 1 / C\ i$

Please note that the current is sinusoidal with the same frequency and we want to find the gain and phase shift.

Using

$I = B \cos(wt+ \varphi)$

Using the system equation

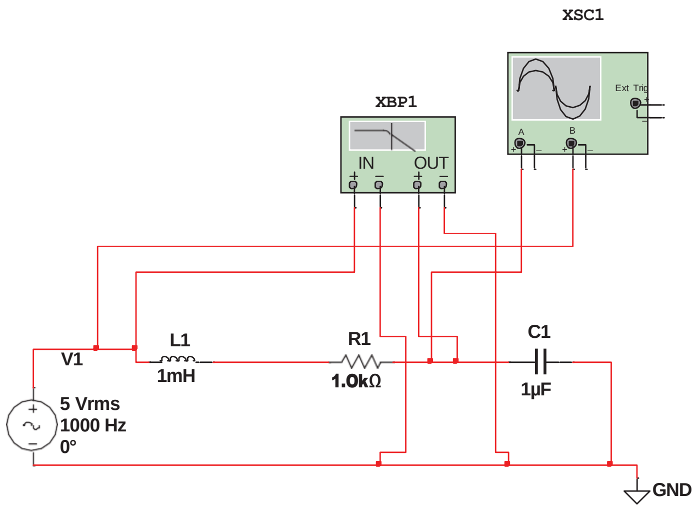
$$A w \cos(wt+90) = B L W^2 \cos(wt+180 + \varphi)$$
$$+ B R\ w \cos(wt+90+\varphi)$$
$$+ B / C \cos(wt+\varphi)$$

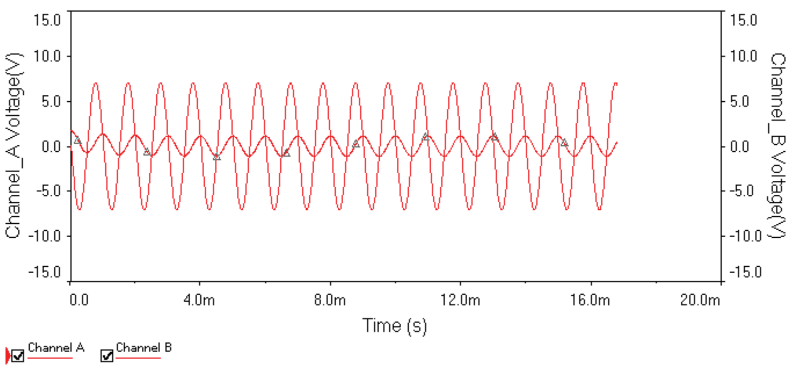
Putting in phase form

$Awj = - B L w^2$ angel $\varphi + j$ RBw angel φ + B/C angel φ

B angel φ = $Awj / (-L W^2 + Rwj+1/C)$

Please note that for sinusoidal input voltage what matter is the phase deference between inputs and output current so to simplify things we take the input voltage phase as zero.


XSC1
Ext Trig
A
B
XBP1
IN
OUT
V1
L1
1mH
R1
1.0kΩ
C1
1μF
5 Vrms
1000 Hz
0°
GND


15.0
10.0
5.0
0.0
-5.0
-10.0
-15.0
Channel_A Voltage(V)
Channel_B Voltage(V)
0.0
4.0m
8.0m
12.0m
16.0m
20.0m
Time (s)
Channel A
Channel B

When we have two negative real poles p1 and p2 the solution is

$$Y(t) = C1e^{-p1t}+C2e^{-p2t}+C3$$

We can find C1 and C2 and C3 from initial conditions

When we have imaginary poles

$$P1=\alpha + jwn$$

$$P2= \alpha - jwn$$

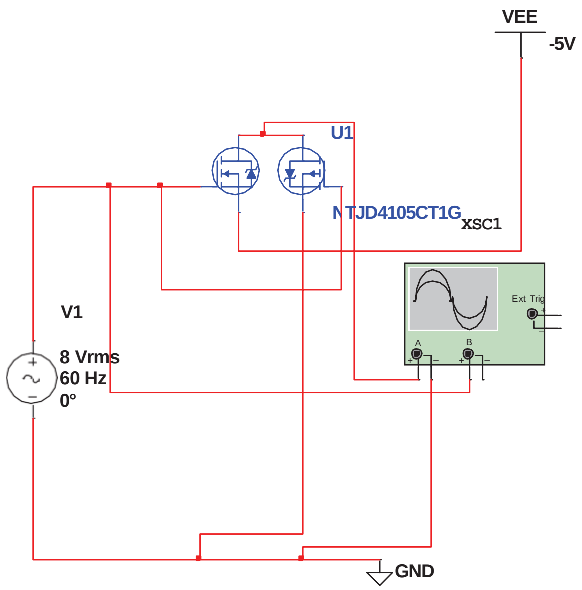
The solution is

$$Y(t) = C1\ e^{-\alpha t}\ (e^{j(wt+\beta)} + e^{-j(wt-\beta)})$$

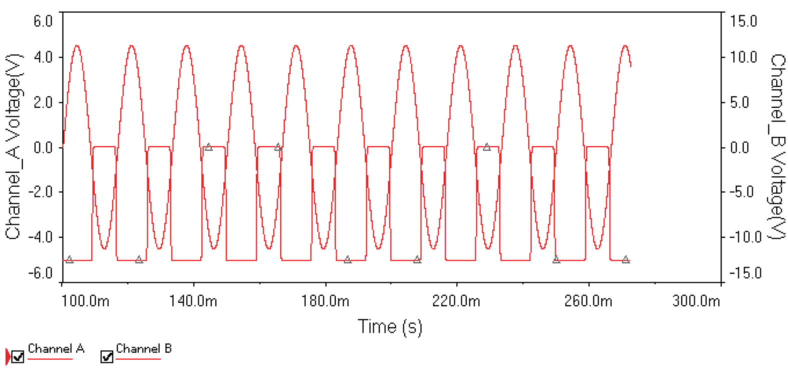
This is attenuation as a function of time and the attenuation factor is α, and oscillation with natural frequency wn.

We have an example for the two negative real poles and the two complex conjugate poles.

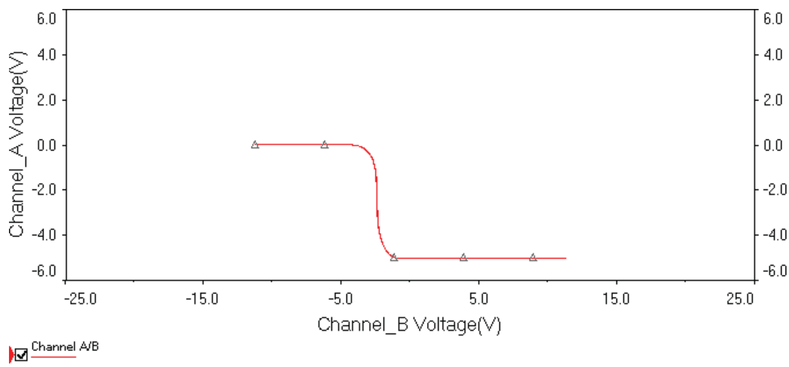
CMOS logic

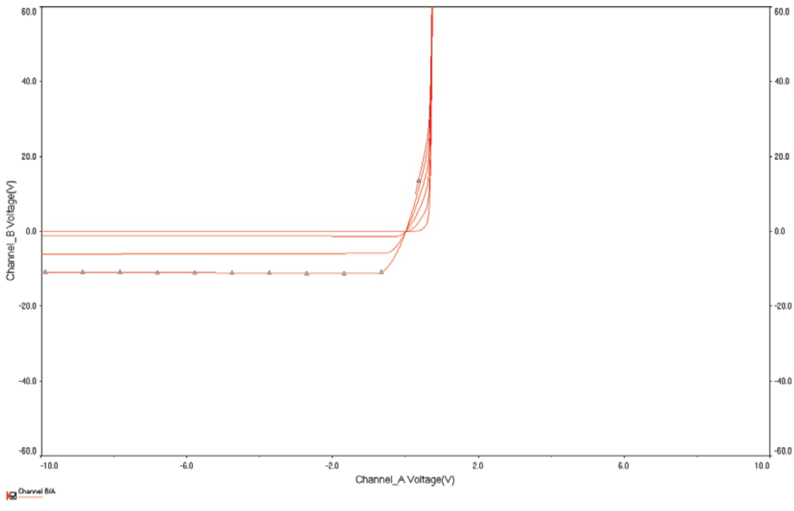


VEE
-5V
U1
NTJD4105CT1G
XSC1
Ext Trig
A
B
V1
8 Vrms
60 Hz
0°
GND



6.0
4.0
2.0
0.0
-2.0
-4.0
-6.0
Channel_A Voltage(V)
15.0
10.0
5.0
0.0
-5.0
-10.0
-15.0
Channel_B Voltage(V)
100.0m
140.0m
180.0m
220.0m
260.0m
300.0m
Time (s)
Channel A
Channel B





We use CMOS to build logic for computers. CMOS is good because of low power consumption. This is due to fast transition from low to high or high to low.

In CMOS inverter we have two transistors. One is npn and the other is pnp. We also have Vcc and ground. The bases of the two transistors are connected together.

The circuit is connected as in figure 1. If the input to this circuit is low (zero volt) the output is high (five volts) and the opposite is true.

The fact that transistor can be used as a switch make it possible to build digital computers. Not only inverters but also all logic gates can be built using CMOS. These logic gates are the basic building blocks for digital computers.

We can explain the operation of CMOS inverter simply by switches. For logic high (five volts) the lower transistor is short and the upper is open so the output is zero. For logic low (zero volt) the upper transistor is short connecting the output to the five volts supply and the lower transistor is open so the output is high (five volts).

This is the operation as ideal switches. Now we are going to increase the voltage from low to high in a linear way and look at the output of the inverter as a function of this input. We do this to obtain the characteristic curve of the CMOS inverter. This curve will explain the advantage of CMOS technology. We will see from this curve that the transition from high to low is very sharp which explain the low power consumption.

Now we will study the case from operating point view. Graph four show the characteristic curves for npn transistor.

Let us take the ideal case were input low is zero and input high is five volts and output low is zero and output high is five volts. In this case as we increase the voltage in a linear manner one characteristic curve will be increasing and the other will be decreasing and the two are mirror image around 2.5 V except for the shift up or down.

If we have this, the operating point will be low for input 0 < Vin < 2.5 and high for input 2.5 < Vin < 5. The result will be curve three.

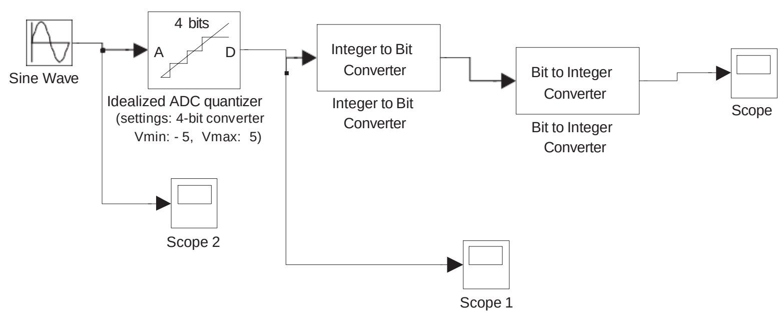
One transistor is npn MOS and has a pinch off voltage at one volt. This mean as we increase the input voltage from zero the current will stay zero until one volt then it will increase according to the square law.

The second transistor is pnp MOS with characteristics as the mirror image around 2.5 volts of the npn MOS.

It is important to have the 2.5 mirror image symmetry so that the transition happens at 2.5 volts.

Let say we have a mike connected to a ADC. The ADC is 8 bits and the dynamic range from −4 volts to 4 volts. The 8 bits is put in an 8 bit register and transmitted serially from the LSB to the MSB bit by bit to another 8 bits register.

Timing is very important in this operation. We transmit bit by bit at a certain constant rate. This rate is bit clock. We need also a word clock. This clock is from the beginning of the word to the end of the word.



The first graph is a sin wave. This wave will be the input to a ADC. The DAC is 3 bits so the dynamic range from −5V to 5V will be quantized to 8 levels. In PCM we send bit by bit the three bits of the quantized value serially starting by the MSB. The time for each bit is the sampling time divided by three.

The receiver is going to receive these three bits and give us the analog value to this number. This can be a shift register which input bit by bit and when we have the three bits a DAC give us a real number.

First we specify the dynamic rang. In this case it is −5 volts to 5 volts. Second we get the number of levels by knowing the number of bits in the register. In this case the three bits give 8 levels.

Know we quantize the dynamic rang to levels were 10/8 is the size of the level.

Each level is given an integer in increasing order one to 8 were all analog inputs between one level and the next is given this value of the integer.

Know this integer in decimal is converted to binary and transmitted bit by bit to the receiver.

Know let us talk about the hardware. The ADC can be a counter converter or a flash converter.

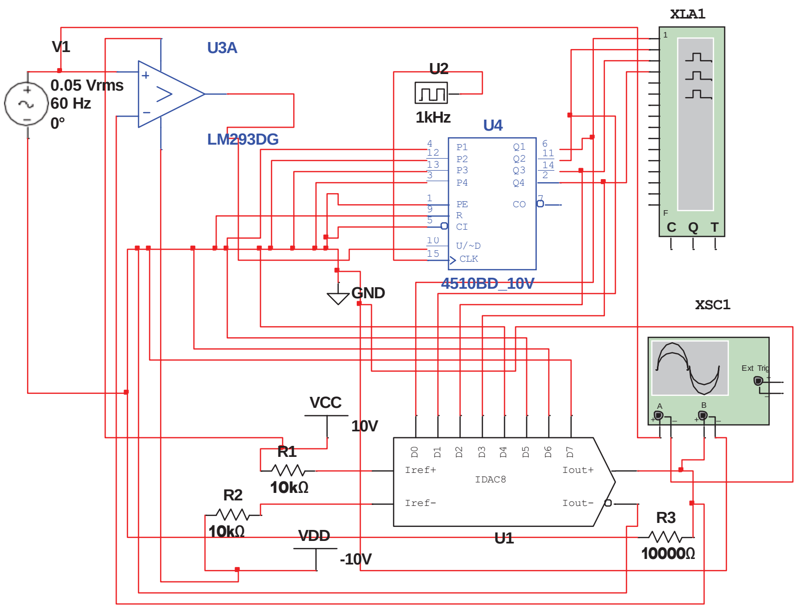
We will talk about the counter converter. Let us say that we have the output of the counter in analog and we compare this analog output with the input signal by a comparator.

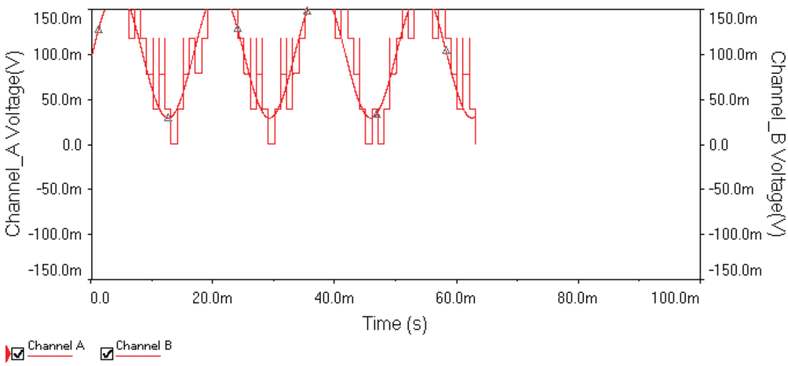
If the input of the system is higher than the analog output, the comparator gives a high signal to the counter to count up. If the input signal is lower than the output signal in analog, the comparator will give a low signal to the counter to count down. This way we will track the input and the counter register will have the input value in digital.

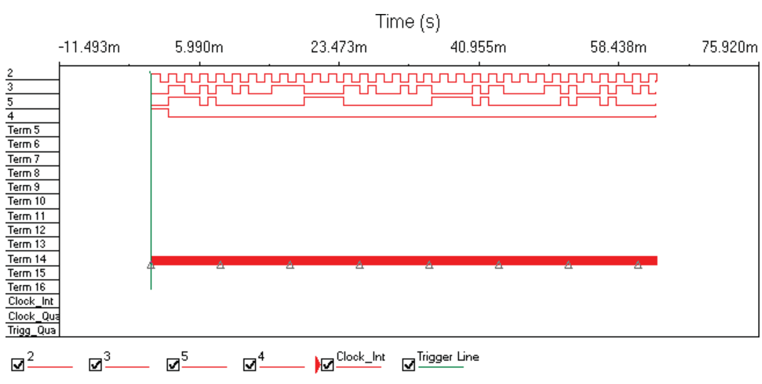
Note that the LSB is delta modulation of the input wave.

Another kind of converters is the flash converter. As you can see the counter converter is not instantaneous. It will take some time to track the input. On the other hand the flash converter is instantaneous. We give it an analog input it will give us the digital output in one clock.

It has a resistive part and a logic converter part. The resistive part is equal resisters in series and constant input voltage.




150.0m
100.0m
50.0m
0.0
-50.0m
-100.0m
-150.0m
Channel_A Voltage(V)
Channel_B Voltage(V)
0.0
20.0m
40.0m
60.0m
80.0m
100.0m
Time (s)
Channel A
Channel B


Time (s)
-11.493m
5.990m
23.473m
40.955m
58.438m
75.920m
2
3
5
4
Term 5
Term 6
Term 7
Term 8
Term 9
Term 10
Term 11
Term 12
Term 13
Term 14
Term 15
Term 16
Clock_Int
Clock_Qua
Trigg_Qua
2
3
5
4
Clock_Int
Trigger Line

Generation of random variable for a given PDF

In simulation software sometimes we need white noise signals. In this work I will try to generate white noise in a digital way. We need a sin wave and a sampler and a quantizer and an adder to do this work. Let us say that the sin wave is quantized to more than ten levels and the sampling of the sin wave is not periodic with the wave. We need the number of samples to be large comparing to the wave period. Then we need to quantize each sample. Then we need to calculate the quantization error for each sample by subtracting the sin wave value at that sample from the quantization level for that sample. If we look at the error we will find that the error values are equally likely distributed from $-q/2$ to $q/2$.

At this point we have a function of samples that are equally likely distributed. How can we use this to get a function of samples that are normally distributed? In probability if we add two random variables the PDF of the sum is the convolution of the two PDFs. If we are able to generate three random variables with three widths that are proportional to one, two and three we can use them as a base to estimate any given PDF [16].

This story is like the story of Maxwell and Hertz. Usually data is given and we want to generate the PDF. In this case the PDF is given and we want to generate the data.

We can estimate any PDF with a good accuracy using the three generated PDFs as a base. For example, we can generate a triangular PDF by adding two random variables that have equally likely PDFs.

As we can see we can use the three equally likely PDFs with three chosen time constants (the width of the rectangle) to estimate any given PDF. Say we take the PDF to the frequency domain by taking the Fourier transform. By Knowing the places of the poles of the Fourier transform we can design the time constants for a given PDF.

Let me try to explain. We can generate three equally likely PDFs with three different widths T1,T2 and T3. I say that this can be used as a base to generate any given PDF. To do this we have to take the given PDF to the frequency domain. This can be done by taking the Fourier transform of the given PDF that we want to estimate or generate data for. By looking at the frequency domain we can chose the places of the poles that estimate this frequency response. Each pole is related to one T and by knowing the three poles we can find T1, T2 and T3.

But how can we generate the base? This can be done by changing the quantization error. To change the quantization error we need to change the quantization levels. For example we can double the error by decreasing the levels from eight to four. This is for the sin wave. Let us say it is four volts peak to peak.

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